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DMS-100 Family and SL-100

DMS VoiceMail

Administration Guide (for single-customer systems)

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- Chapter 5 - “Making recordings”
  - The call answering greeting (page 5-2).
- Chapter 8 - “Hardware Administration”
  - Modifying T1 channels (page 9-9).
- Chapter 10 - “System Status and Maintenance”
  - SMDI Link Status (page 10-24).

This version make all previous versions obsolete.

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About this document

This document details the administration procedures to be performed by the DMS VoiceMail system administrator.

When to use this document

This document is written for DMS-100 Family offices with a Service Peripheral Module (SPM). More than one version of this document may exist. To determine whether you have the latest version of this document, check the release information in *DMS-100 Family Guide to Northern Telecom Publications*, 297-1001-001.

How DMS VoiceMail documentation is organized

This document is part of DMS VoiceMail documentation that supports the Northern Telecom line of DMS VoiceMail products. DMS VoiceMail documentation is a subset of the DMS-100 Family library.

The DMS-100 Family library is structured in numbered layers, and each layer is associated with an NT product. To understand DMS VoiceMail products, you need documents from the following layers:

- DMS-100 Family basic documents in the 297-1001 layer
- DMS VoiceMail documents in the 297-7001 layer

DMS VoiceMail documents and other documents that contain related information are listed in “Finding DMS VoiceMail information” in *DMS VoiceMail* Product guide. (297-7001-010)

References in this document

The following documents are referred to in this document.

Number	Title
297-7001-100	*DMS VoiceMail Planning and Engineering Guide*
297-7001-310	*DMS VoiceMail Translation Procedures*
297-7001-501	*DMS VoiceMail Routine Maintenance Procedures*

Number	**Title**
297-7001-503	*DMS VoiceMail Trouble-locating and Alarm-clearing Procedures*
297-7001-510	*DMS VoiceMail Maintenance Messages (SEER) manual*

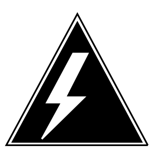
What precautionary messages mean

Danger, warning, and caution messages in this document indicate potential risks. These messages and their meanings are listed in the following chart.

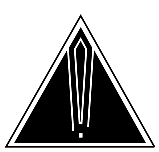
Message	Significance
DANGER	Possibility of personal injury
WARNING	Possibility of equipment damage
CAUTION	Possibility of service interruption or degradation

Examples of the precautionary messages follow.

DANGER
Risk of electrocution
The inverter contains high voltage lines. Do not open the front panel of the inverter unless fuses F1, F2, and F3 have been removed first. Until these fuses are removed, the high voltage lines inside the inverter are active, and you risk being electrocuted.



WARNING
Damage to backplane connector pins
Use light thumb pressure to align the card with the connectors. Next, use the levers to seat the card into the connectors. Failure to align the card first may result in bending of the backplane connector pins.



CAUTION
Loss of service
Subscriber service will be lost if you accidentally remove a card from the active unit of the peripheral module (PM). Before continuing, confirm that you are removing the card from the inactive unit of the SPM.

Typographic conventions

The following conventions are used throughout this guide:

- ***Softkeys* -** are displayed on the various administration menus and screens and indicate which keyboard function keys carry out specific DMS VoiceMail tasks. These are referred to in the document by using the label of the softkey (as displayed in the given menu), delimited by square brackets.

  Examples: [Exit], [OK to Delete], [Save]

- ***Keyboard keys* -** (or hardkeys) are referred to by indicating the label of the key, delimited by angle brackets.

  Examples: <1>, <2>, <Return>

- ***Text input* -** Where you are required to input specific text, the characters are presented in bold instead of using angle brackets.

  Examples: **servord**, **custpwd** (not <s><e><r><v><o><r><d>)

- ***Fields in administration screens* -** When the name of a field is referred to, it appears in italics and in a different typeface than the body of the document.

  Example: Enter a unique identifier in the *Announcement ID* field.

- ***Values in fields* -** When the choices presented in a selectable data field are discussed, they are in quotes.

  Examples: The default is "Enabled".
  Select "Custom" to create a set of restriction/permission codes unique to this thru-dialer.

- ***Spoken words* -** Suggested wordings for prompts (such as for voice menus or voice forms), or words which you may be required to speak into the telephone receiver, are in italics and between double quotation marks.

  Example: An appropriate prompt would be *"Please wait on the line, an attendant will be with you shortly"*.

References

In this manual, where reference is made to another part of the manual, or to another document, the following conventions are used:

- References to section headings and chapter titles are surrounded by double quotation marks.

  Examples: See the section "Deleting voice menus" later in this chapter.
  See "Time-of-Day Controls" in the "V oice Administration" chapter.

- References to other NTPs or documents are in italics.

  Example: See the *Translations Guide* (NTP 297-7001-310) for details.

Understanding DMS VoiceMail administration

This chapter includes a description of the capabilities and operation of DMS VoiceMail, the relationship of DMS VoiceMail to the telephone network, and an overview of the DMS VoiceMail administration process.

DMS VoiceMail

DMS VoiceMail is a voice processing system designed to provide call answering and voice messaging services to the Central Office (CO) as well as the private business (CPE or Central Premise Equipment). A DMS VoiceMail system consists of a Service Peripheral Module (SPM) and voice processing software, and is administered from a local or remote terminal.

> ***Note:*** In a CPE environment, the Service Peripheral Module is referred to as the Message Services Module (MSM) and DMS VoiceMail is known as Meridian Mail. However, throughout the rest of this guide, only the terms SPM and DMS VoiceMail will be used.

In a CO environment, DMS VoiceMail supports the DMS-100 switch as well as other Central Office switches that support the Simplified Message Desk Interface (SMDI) that conforms to the applicable Bellcore standard. CPE indicates one of the following situations:

- A private business maintains its own SL-100 and MSM on site. In this situation, both the SL-100 and Meridian Mail are administered by the customer.
- A private business does not have its own switch but does have its own MSM on site. In this circumstance, the business is a centrex customer, but maintains its own voice messaging system (MSM) on its premises. This is known as CPE Centrex.

DMS VoiceMail provides a variety of voice mail services which are sold to subscriber groups as packages. A package may include some or all of the available services.

DMS VoiceMail subscribers are assigned a voice mailbox which they access using a private password. Recorded prompts guide users whenever necessary, and also assist callers to leave messages.

The Call Answering feature package includes call answering and message retrieval functions, with a subset for subscribers with dial pulse sets. It is primarily intended for residential and small business subscribers. It can, however, be used to provide simplified call answering services to users within private networks.

The Voice Messaging feature package offers enhanced voice mail capabilities in addition to basic call answering and message retrieval. This feature is primarily intended for private businesses and Centrex users.

Optional feature packages include AMIS Analog Networking, Voice Forms, Voice Menus. The following packages may also be purchased optionally for CPE systems only: Meridian Networking, Access, and AdminPlus.

Voice Messaging and Call Answering

Your DMS VoiceMail system can have one of two features installed: Voice Messaging or Call Answering. The feature that is installed determines the screens and fields that are displayed in the administration screens. It also determines the features and the telephone interface that are available to users on the system.

Voice Messaging

The Voice Messaging feature provides users with full voice messaging functionality in addition to call answering and message retrieval capabilities. In a Central Office (CO) environment, it is primarily intended for Centrex customers. In a private network or business environment (CPE), full-featured voice messaging is typically provided to users. Voice Messaging provides users with the following:

- ***call answering functionality*** which allows callers to leave messages for users who are away from their phone or on the phone.
- ***voice messaging functionality*** which allows users to:
  - compose and send messages to other local voice users (or users at other DMS VoiceMail sites, if Networking is installed);
  - reply to the sender of a message with a keypad command;
  - immediately call back the sender of a message with a single-digit keypad command;
  - directly deposit messages into another user’s mailbox without first ringing their extension;
  - forward messages to other users;
  - dial another user by name instead of their extension; and
  - tag messages as urgent.

- ***advanced features*** which allow users to be notified of new messages at a remote phone or pager, send messages to non DMS VoiceMail users, and create personal distribution lists.

MMUI

Voice Messaging uses the Northern Telecom proprietary interface MMUI (Meridian Mail User Interface). This is the telephone interface that provides users with access to voice messaging features such as Compose and Send. When Voice Messaging is enabled, a field in the Voice Messaging Options screen is automatically set to MMUI.

See Chapter 3 in the *Product Guide* (297-7001-010) for a list of V oice Messaging features.

Call Answering

The Call Answering feature (also known as Simplified Call Answering) provides users with a simplified interface. As a result, users have access only to basic call answering and message retrieval capabilities. This means that a caller can leave a message for a user who is away from, or busy on, the phone. Unlike voice messaging users, call answering users do not have access to voice messaging functions (i.e., they cannot Compose and Send voice messages).

In a Central Office (CO) environment, Call Answering is primarily intended for residential and small business customers. In a business (CPE) environment, it can be used for users who do not require full-featured voice messaging functionality. For example, in a university setting, you may not want students to compose and send messages, yet you want them to have access to call answering functions.

VMUIF

Call Answering uses an interface style that is compatible with recommendations of the Voice Messaging User Interface Forum (VMUIF). (Throughout this NTP, this interface is referred to as VMUIF.) This is a simplified telephone interface that provides users with call answering and message retrieval functions. When Call answering is enabled, a field in the Voice Messaging Options screen is automatically set to VMUIF.

See Chapter 3 in the *Product Guide* (NTP 297-7001-010) for a complete list of Call Answering features.

Users and subscribers

In a business (CPE environment) end users are referred to as *users* whereas in a central office, end users are typically referred to as *subscribers* since they subscribe to certain features that are offered by the service provider. Both terms are used throughout this guide since it is intended for both CO and CPE environments. However, in the administration screens, the term users is employed exclusively.

System capacity

The number of mailboxes on a DMS VoiceMail system is calculated by the total available hours of storage, divided by the average time taken by each user's messages and greetings. The average per mailbox time depends on the mailbox size limits and message deletion policy, both of which are set by the service provider.

The Service Peripheral Module is provisioned by selecting appropriate numbers of voice ports and hours of storage. The amount of memory is fixed and is sufficient to run all the supported applications and utilities under full load even in the presence of single point failures. Capacity will be limited more by the number of ports than by limitations of the Service Peripheral Module.

Table 1-1 shows current maximum capacities for systems with 192 ports. Some of these capacities may not be applicable to certain configurations.

Table 1-1xxx
DMS VoiceMail system capacities

Item	Maximum
Voice messaging channels	192
Voice storage hours	1200
Storage hours for voice menus, voice form definitions and personal verifications (spoken name)	100
Customer groups per system	2000
Registered mailboxes per system	42,672
Message per mailbox	999
Minutes per mailbox	360
Voice service DNs	4000
Voice menus	4000
Networking nodes per system	50
Distribution lists per organization	no limit
Entries per organization list	120
Distribution lists per mailbox	9
Entries per mailbox list	99
Administrative positions	4
Maintenance console	1
Maintenance printer	1
SMDI links (8 redundant or 16 non-redundant)	16

Performance standards

Under normal conditions, for most voice messaging functions, response time should be under one second 95% of the time, and over four seconds no more than one per 10,000 instances.

Administration of DMS VoiceMail

The administration and maintenance interface for the Service Peripheral Module can be monitored by the service provider either locally or remotely.

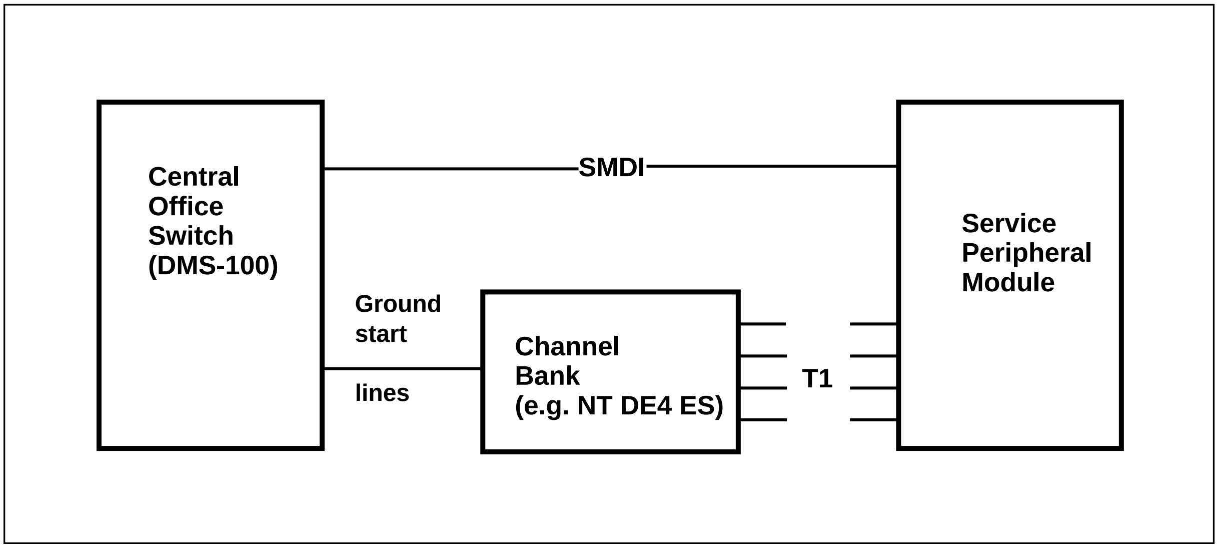
Up to four administrative positions can operate simultaneously from locally or remotely attached terminals. Allowed are one main administration terminal and up to three user administration terminals (UATs) which can only be used to perform user administration. System events are recorded in a log file and reports are printed on a locally attached printer.

The system can be administered remotely via modem. However, the system cannot be administered both locally and remotely at the same time. Local versus remote administration is a toggle.

DMS VoiceMail in the Central Office environment

The Service Peripheral Module for DMS VoiceMail is a voice processing server developed for DMS-100 and other central office environments. It contains up to 192 voice channels for the service provider to provide voice mail service to subscribers. Figure 1-1 shows how DMS VoiceMail can be integrated in a CO environment.

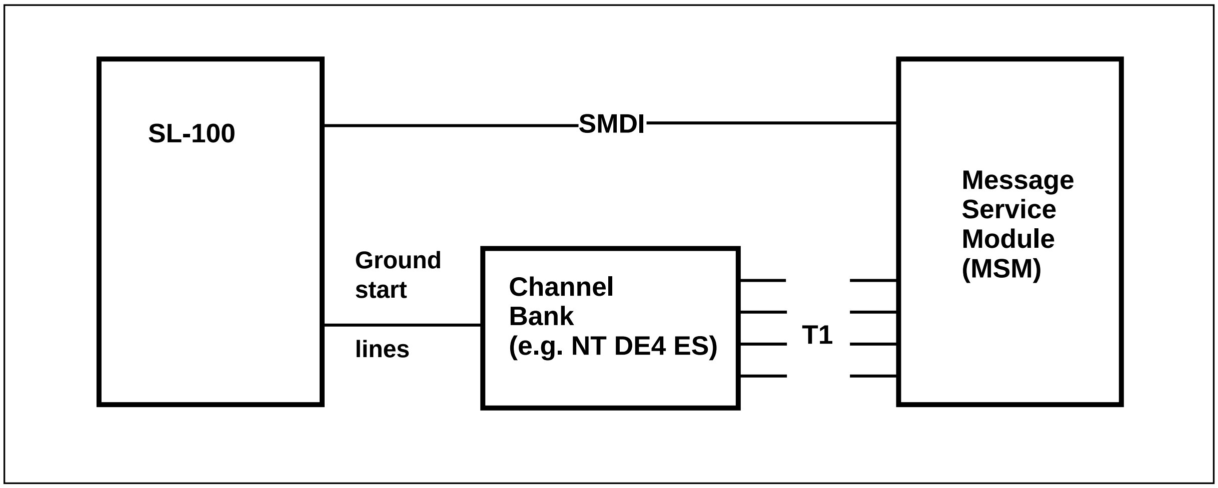
Figure 1-1xxx
DMS VoiceMail in a Central Office (CO) environment



Meridian Mail in the CPE environment

The Message Service Module (MSM) for Meridian Mail is a voice processing server developed for the CPE environment. The MSM has a capacity of 192 voice channels.

Figure 1-2xxx
Meridian Mail in an private or business (CPE) environment



Contents of this guide

This manual describes the basic administration tasks that you will carry out on your DMS VoiceMail system. It assumes that all the hardware, including the administrator's terminal and optional printer, is in place. DMS VoiceMail administration facilities are used in the initial setup of your system as well as for routine maintenance.

Additional administrative tools and utilities are available.

This guide covers the following topics:

- ***Administrative role and responsibilities*** - Your role and responsibilities as administrator are covered in this chapter and in the chapter "Setting up the system".
- ***Procedures for setting up and administering the system*** - If you are setting up DMS VoiceMail, read the chapter "Setting up the system" before commencing with any of the procedures described in this guide. When setting up for the first time, certain procedures need to be performed before others. This chapter explains this order and points out those parameters that *must* be configured. Procedures required to set up and maintain the DMS VoiceMail system are described throughout this guide. This includes basic setup procedures, some maintenance procedures (such as backing up the system), Voice Menu applications,

procedures carried out using the telephone (recording personal verification greetings, announcements and a custom call answering greeting) and procedures for administering optional features such as Networking.

Note: Your system may not include all of the features described in this guide. To obtain features that you do not have, contact your sales representative.

Organization of chapters

The division of this manual reflects the hierarchical set of procedures accessible from the Main Menu. Each item that appears in the Main Menu has a corresponding chapter describing the administrative tasks, and the screens and fields one interacts with to complete the tasks. Each screen and sub-screen in the DMS VoiceMail administrative facility is described using the following structure:

- ***Introduction*** - a brief description of the menu, and any concepts or rules necessary to use the menu
- ***Menu*** - an illustration of the menu and its softkeys.
- ***Screens -*** an illustration of the screen and its softkeys.
- ***Field descriptions*** - a description of each field as it appears on the screen, stating requirements your entries must meet and any default information supplied by the system.
- ***Choice of Actions -*** A description of available softkeys and their actions.
- ***Task-oriented Procedures*** - are step-by-step descriptions of administrative tasks. They are provided when additional steps are required to complete a task (i.e., in addition to filling in the described fields and using the softkeys).
  - ***Starting point*** - tells you where in the menu hierarchy the procedure begins.
  - ***Body of procedure*** - is a numbered list of the required steps and any additional information you may require to complete a task.

An overview of administration

As administrator of DMS VoiceMail, your functions include setting up the initial system configuration (normally a once-only operation) and performing some routine procedures needed for effective operation of the system. Before proceeding with the initial setup of your system, review the *DMS VoiceMail Planning and Engineering Guide* (NTP 297-7001-100). This document provides guidelines for planning and preparing information that is required during the initial setup of your system.

If you are working with an engineering organization to set up your system, you may also have to deliver collected data related to the performance and use of the DMS VoiceMail system (Service Peripheral Module). Your role as administrator in supporting engineering is to review and analyze the data to identify early indications of resource shortages. This data is used in office provisioning calculations. In addition, the administrator collects and supplies data to the maintenance organization for detecting equipment faults.

Administrative procedures are performed either through easy-to-follow, menu-driven screens at your administration terminal or through your telephone. You may need to carry out some procedures frequently, perhaps daily, others only occasionally.

At the DMS VoiceMail administration terminal

The setup and operation of your DMS VoiceMail system involves work at the main administration terminal. (You cannot use one of the secondary user administration terminals since they only give you access to user administration.)

Through the administration terminal you can access the screens and menus used to define the characteristics and parameters of your system. Each chapter in this manual describes procedures carried out at a particular menu or set of screens.

System administration can be broken down into the following categories:

- ***User Administration*** - involves the maintenance of a current information base of users, user models, and system distribution lists, and carrying out other user-related functions such as recording personal verifications for users.
- ***General Administration*** - involves setting general system parameters (such as the attendant DN and SEER print port name); backing up the system from the hard disk onto tape; changing the system administrator password and system time.

  Note: If you have the Multiple Administration Terminals feature, up to four terminals can be configured (one main administration terminal and three secondary terminals). The secondary terminals can only be used for user administration and can either be local or remote.

- ***Voice Administration*** - involves the administration of all voice services used by your organization, including dedicating voice channels to a particular service, assigning phone numbers to voice services, setting operational parameters and security for voice services and administering voice services such as Voice Menus, Outcalling (Remote Notification and Message Delivery to Non-users) and Voice Forms.
- ***Hardware Administration*** - involves viewing the contents of the hardware database for your DMS VoiceMail system. This will give you an idea of the number of nodes in your system, and the type of cards and ports that have been configured for those nodes. You can also print node and data port information while in the Hardware Administration menu. Hardware Administration does not, however, involve modification of the hardware database. The database is modified using System Administration Tools, described in Appendix A.
- ***System Status and Maintenance*** - involves monitoring the operational status of the system, including System Event and Error Reports (SEERs) for use in troubleshooting. Should any components, such as nodes or cards, require servicing, this also involves disabling those components prior to servicing.
- ***Operational Measurements*** - involves setting operational measurement options and viewing system usage statistics. This information is presented to you in the form of various reports, such as traffic reports for various features (voice messaging, voice menus, networking, AMIS networking and outcalling), disk usage reports, DSP port usage reports and user usage reports.
- ***Network Administration*** - involves the administration of one or more of Meridian Networking and AMIS Networking. Each of these networking components is optional and may be combined in various ways. Networking allows one DMS VoiceMail system to communicate with other DMS VoiceMail systems. AMIS Networking allows DMS VoiceMail users to send messages to and receive messages from users of other voice messaging systems subscribing to the AMIS protocol (which may include non-DMS systems).

At the telephone

To create the various voice recordings required for your system, you must use a telephone as well as the administration terminal. The basic procedures for creating voice recordings are described in detail in the chapter “Making recordings”. You may create the following types of voice recordings:

- ***Personal Verification Recordings*** - A recording of a person’s name (and extension) may be recorded for each user. When recorded, it is played to callers instead of the user’s phone number, making identification easier. Personal verifications can either be recorded by the administrator at the administration terminal, or by users with their telephone sets.

  Note: Personal verifications are not necessary for Call Answering subscribers.

  Verifications can also be recorded for DMS VoiceMail network sites. If no verification is recorded, a recording of the site number is played when callers are connected to a remote user’s mailbox to leave a message. This is used to identify the site. If a personal verification has been recorded, the site name is played instead, making identification easier.

- ***Custom Call Answering Greeting*** - This greeting is played to external callers who reach the call answering service and is simply a recording of the customer’s name. It is played before any personal greetings.

  Note: This greeting does not apply if Call Answering (VMUIF) is installed on the system.

- ***VMUIF Introductory Tutorial Greeting*** - This greeting is played to Call Answering subscribers the very first time they log on to their mailbox. It describes how to use the call answering system and the features that are available.

  Note: This greeting does not apply if Voice Messaging (MMUI) is installed on the system.

- ***Broadcast Messages*** - A broadcast message is deposited in the mailboxes of all DMS VoiceMail users on the system.
- ***Voice Prompt Maintenance*** - The routine recording of prompts, announcements, and greetings used in various voice services.

Administration overview

For a better picture of what your administrative responsibilities are and how they relate to each other, Figure 1-3 illustrates a conceptual view of administration and Figure 1-4 illustrates the hierarchy of menus available at the administration terminal.

Figure 1-3xxx
Administration overview

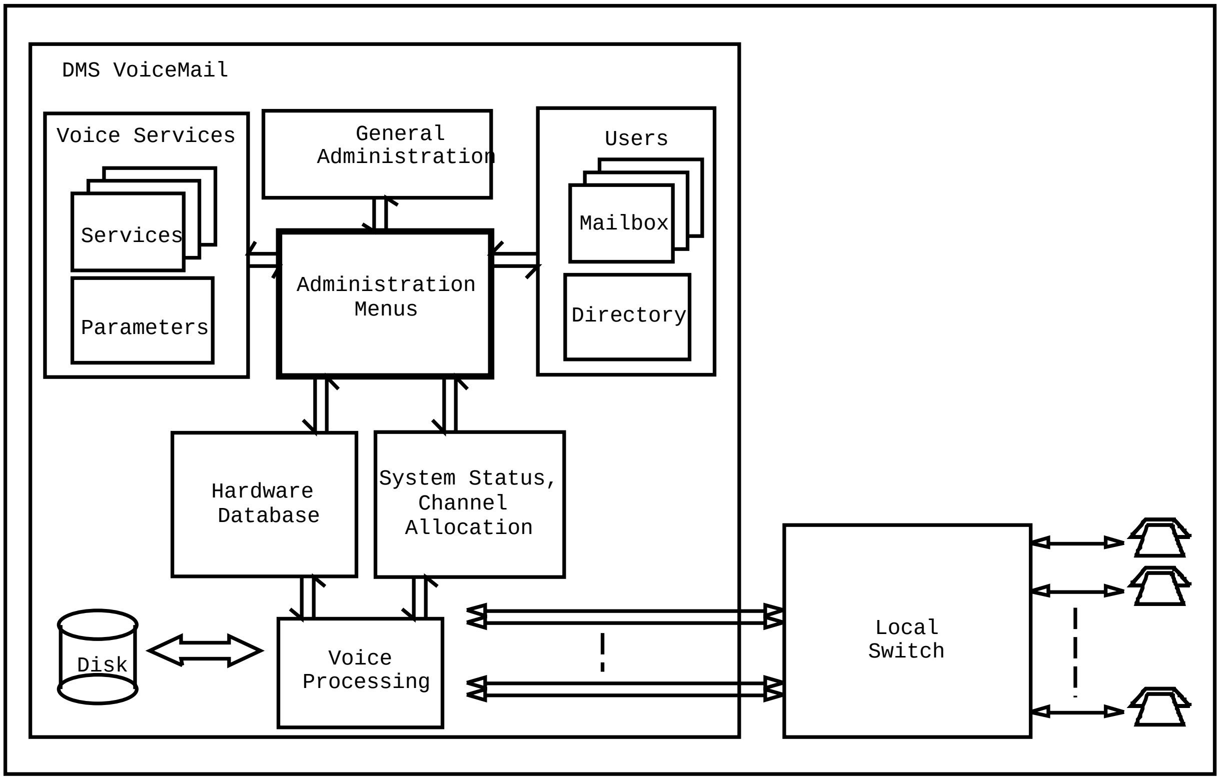
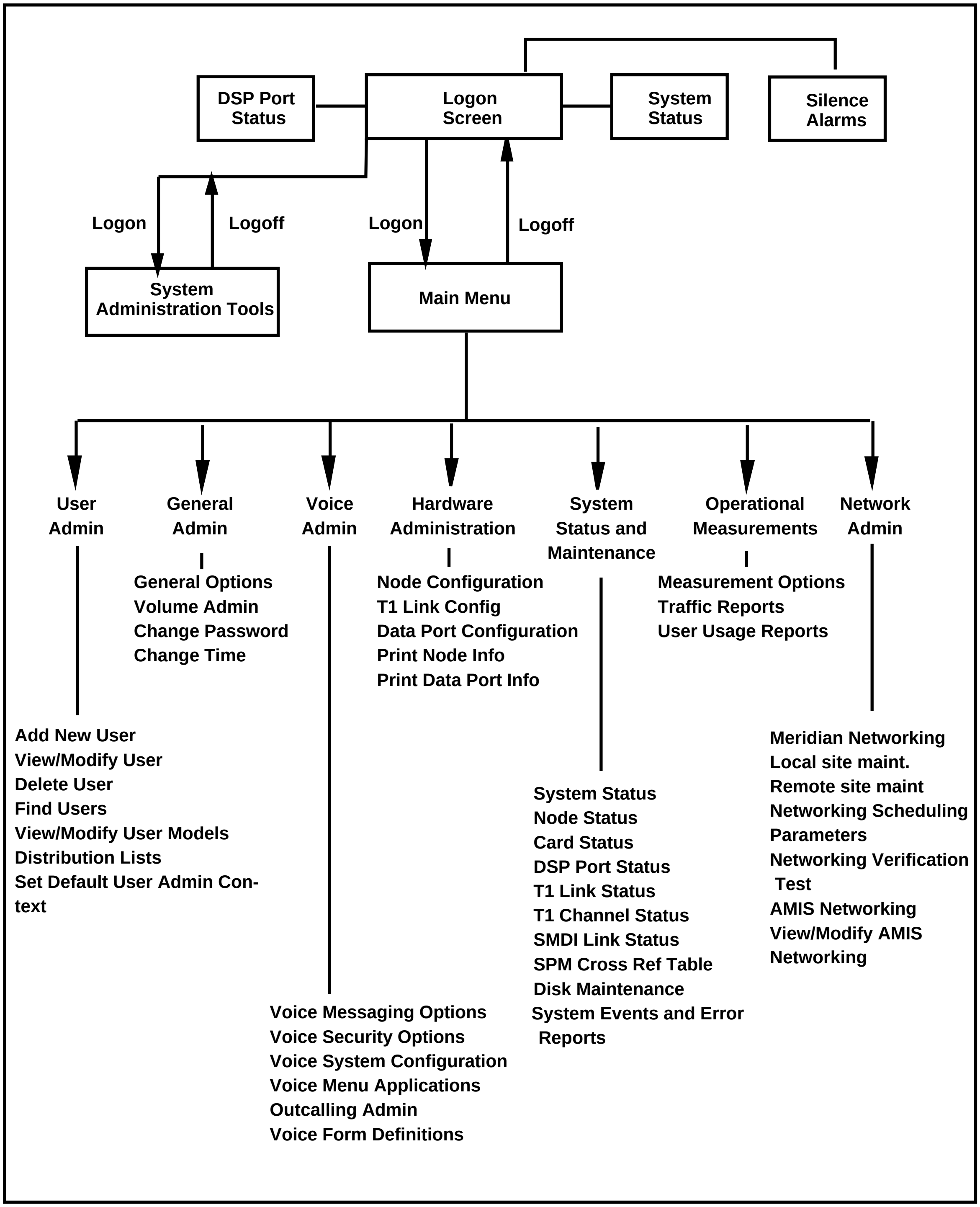


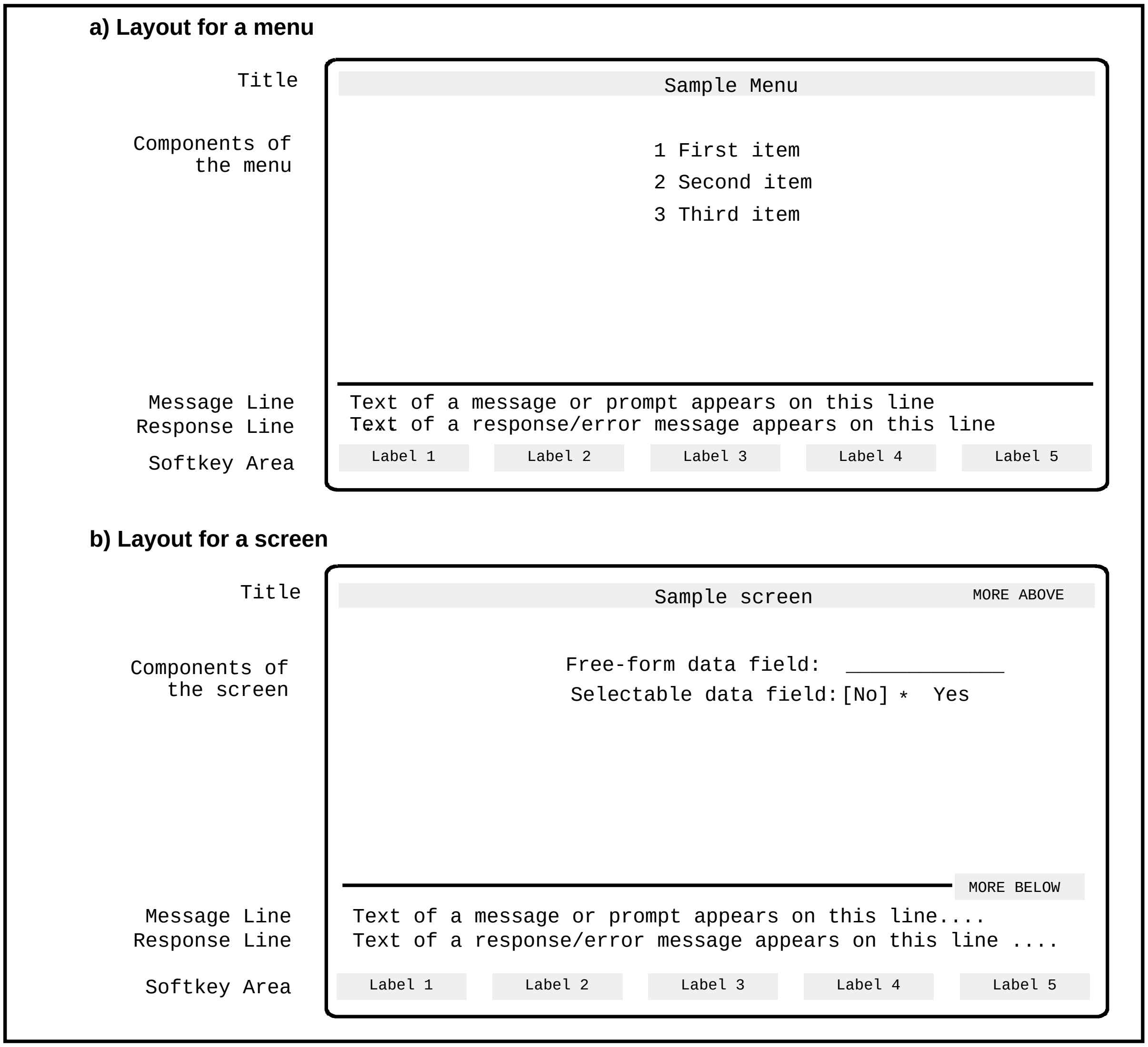
Figure 1-4xxx
System Administration menu hierarchy



System Administration screens: menus and screens

Menus and screens in the System Administration facility conform to the general layout shown in Figure 1-5. The title of each screen or menu appears on the first line of the screen. For menus, this is followed by a list of numbered items. For screens, the title is followed by fields for viewing or entering information. The bottom four lines of the screen are reserved for system prompts, responses, error messages, and softkey identification. Two types of fields appear in administration screens: free-form data fields, where you can overwrite existing entries and enter new data; and selection fields, where the system presents a set of options to which the field can be set. Some fields that you can change are filled in automatically by the system. For example, when you add a new user, some of the information fields take on, by default, the values of the last user you added to the system; having some of these fields filled in makes it easier and faster for you to add new users with similar profiles.

Figure 1-5xxx
General screen layout



* In this guide, items surrounded by square brackets indicate a selected option. On DMS VoiceMail screens, selections are actually shaded.

Softkeys

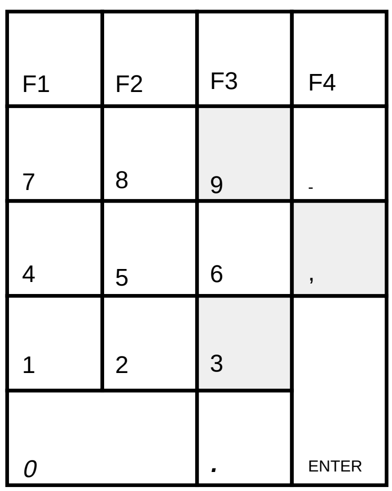
Softkeys appear on the bottom two lines of menus and screens and are displayed in reverse video (dark characters on a light background). They change depending on the menu or screen and may change with the function you are performing. They correspond to function keys F6 through F10 on the top row of the keyboard. They also correspond to the keys on the keypad shown in Figure 1-6.

Keypad functions

Figure 1-6 also shows the other functions that are available on the keypad by pressing the single keys or the key combinations shown.

> ***Note:*** The functions shown in Figure 1-6 are only available if the keypad is in application mode (application mode is the default whenever the system is rebooted). If you choose to work with a numeric keypad (where the numeric keys generate numbers when you press them), then only the F1, F2, F3 and F4 keys retain the functions indicated. The keypad is set to numeric mode through the terminal's set-up function; for details, consult the documentation for your terminal.

Figure 1-6xxx
Numeric keypad function keys




```
0+1 - Start of field
0+2 - End of field
0+6 - Restore default value
of field
F1 -  Softkey 1
F2 -  Softkey 2
F3 -  Softkey 3
F4 -  Softkey 4
1  -  Previous word in field
2  -  Next word in field
4  -  Previous field
5  -  Next field
7  -  Previous page
8  -  Next page
-  -  Delete field contents
.  -  HELP
ENTER - Softkey 5
```

The Help key

On-line Help is available for most of the menus and screens, including the Main Menu. The <Help> key on the keyboard can be used to display information on whatever screen you are working in. If you require help with a screen, press the <Help> key. Alternatively, you can press the period (.) on your numeric keypad (see Figure 1-6). The system will display a screen showing explanations of all the fields on the menu or screen you are working in. When you are done, use the [Exit] softkey on the Help screen to return to the menu or screen you were working in. Figure 1-7 shows an example of the Help screen for the Main Menu.

Figure 1-7xxx
DMS VoiceMail Help example

```
                          DMS VoiceMail Help

  Main Menu
  The Main Menu provides access to all administrative functions
  Softkey Available:
  Logoff          Logs you off the system.
  Description of Menu Items:
  User            To add a new user, modify or delete an existing user,
  Administration   find users or view/change existing User Models.
                  From this menu you also can maintain system
                  distribution lists.

  General         To maintain general system options and perform
  ____________________________________________________________________

  Exit         Using                                     Keypad
               Help                                      Help
```

Multi-page screens

Certain screens may contain more fields than can be displayed at once on the screen. Additional pages are viewed by:

- ***Scrolling*** - If you see “More Below” at the bottom of a screen, or “More Above” at the top of a screen, use the down-arrow key or <Next Scrn> hardkey to view the next page. Use the up-arrow key or <Prev Scrn> to return to the previous screen. When the “More Below” prompt disappears, you are at the end of the screen; when the “More Above” prompt disappears, you are at the top of the screen.

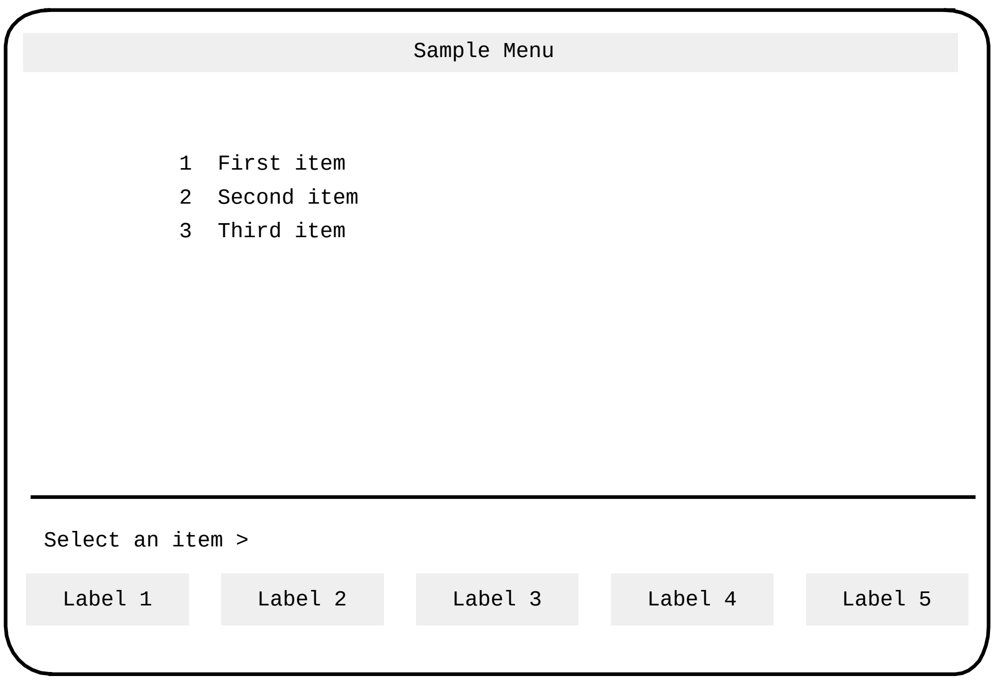
  Note: The down arrow key will only display the last input field, even if there is guide text beyond it. To view any guide text that may appear at the very end of a screen, use the <Next Scrn> hardkey.

- ***Paging*** - Use the [Next Page] softkey if it is displayed.

Selecting a numbered item in a menu

In a menu screen (see Figure 1-8), each item has a number. The system displays a prompt requesting you to select an item. To select a menu item, type the corresponding number and press the <Return> key. The number you enter appears next to the “Select an item >” prompt. When you press the <Return> key, the system displays a sub-menu or screen corresponding to the selected item.

Figure 1-8xxx
Selecting a numbered item in a menu



Entering information in a screen

There are two types of modifiable fields in the DMS VoiceMail administration screens (see Figure 1-9). *Free-form data fields* are fields in which you enter information, such as a user's name or mailbox number. *Selectable fields*, on the other hand, present a series of specific options from which to choose.

In order to modify a field, you must first move your cursor to it. Once the cursor is in the appropriate field, you can change its contents.

Figure 1-9xxx
Entering information in a screen

Some fields display unmodifiable information. You cannot change the content of these fields. The cursor may or may not position on these fields, depending on the type of screen displayed. When a selectable field is not modifiable, only the selected option will be displayed. For example, if a field is disabled, only "No" will be displayed. It will not be shaded.

The following keys on the keyboard and on the application keypad (see Figure 1-6), move the cursor within or across fields:

- <**Tab**> moves the cursor to the next field.
- <**4**> on the application keypad moves the cursor to the previous field.
- <**Return**> moves the cursor to the next field.

- < ↑ > , the up arrow key, moves the cursor to the previous field or the field above.
- < ↓ > , the down arrow key, moves the cursor to the next field or the field below.
- < ← > < → > , the left and right arrow keys, move the cursor in the corresponding direction within an input field, but not between fields. They also move the cursor from one selection to the next in a selectable field.

The following keys change the contents of fields:

- <**Remove**> clears the current field.
- < ⌧ > deletes one character to the left of the cursor each time the key is pressed.
- <**Back Space**> deletes the character on which the cursor is positioned.

Procedure 1-1xxx
Changing the contents of a free-form data field

1 If the field you want to change is below the current cursor position, use one of the following keys to move the cursor to the appropriate field: <Tab>, <Return>, or down arrow key.
or
If the field you want to change is above the current cursor position, use one of the following keys to move the cursor to the appropriate field: up arrow key or <4>.

2 If the field is not blank, delete the current entry using either <Remove> to clear the field, <Back Space> to delete the character on which the cursor is positioned, or < ⌧ > to delete the character to the left of the cursor (until the entry is deleted).

3 Enter the new information.

Procedure 1-2xxx
Changing the contents of a selectable field

1 If the field you want to change is below the current cursor position, use one of the following keys to move the cursor to the appropriate field: <Tab>, <Return>, or down arrow key.
or
If the field you want to change is above the current cursor position, use one of the following keys to move the cursor to the appropriate field: up arrow key or <4>.

2 Use the right and left arrow keys to position the cursor on the appropriate selection.

3 When the cursor is positioned correctly, press <Return> to select.

Selecting an entire line

In some screens you are required to select an entire line. For example, in the View/Modify Directory Entry User screen (Figure 6-11), you must select a name from a list of users to indicate which user profile you want to modify. To select a line in a screen, place the cursor at the beginning of the line and press the <Space Bar>. Screens requiring this mode of selection will indicate this in a prompt (“Move the cursor to the item and press the spacebar to select it”).

Error messages

The system displays error messages, both general and screen-specific, on the line above the softkey display. These messages are simply feedback to the administrator’s actions. (Do not confuse them with SEERS, System Event and Error Reports. SEERs are described in detail in the “System Status and Maintenance chapter”.) The messages remain on the screen until the next user input or until another error message appears. Typical error messages are:

- “The key entered is not valid at this time.”
- “Enter a number in the range of 1 to 6.”

Note: If SEER printing is disabled, reports will print out on the administration screen. To redraw the screen and clean up any interfering information, press <Control> + <r>. This key combination can be used at any time to redraw the screen.

System and user data storage

Each SPM (or MSM) node in the DMS VoiceMail system has a hard disk drive for data storage. The hard disk drives are partitioned into volumes. Volumes are storage areas for system-related or user-related information. The volumes are already set up when the system is installed.

Users are not automatically distributed or balanced among the user volumes. When initially setting up DMS VoiceMail, you must distribute DMS VoiceMail users on the volumes by assigning a given volume number to each user entered in the Add Local Voice User screen. This is described in the section “Distributing users over volumes” in the “User Administration” chapter.

The section “Volume numbers and distribution” in Chapter 7 describes the conventions used for naming volumes and the type of information stored on each volume for the different DMS VoiceMail configurations. At the end of the section, Table 7-2 specifies the storage capacities for each volume.

System security

In today's telecommunications environment every computerized system is potentially open to unauthorized access. DMS VoiceMail has, therefore, been equipped with a number of security features to help minimize this risk.

Ensuring system security

As system administrator, it is your responsibility to take all necessary precautions to prevent security breaches. For example, unless your system has been properly secured, someone who is connected to DMS VoiceMail (such as a user who is logged on to a mailbox or an external caller who has connected to DMS VoiceMail through a call answering session or a voice menu) can place an unauthorized call.

The following fields are located in the Add and View/Modify Local Voice User screens.

- ***Mailbox Thru-dial*** *(also known as extension dialing)* - This feature allows a Voice Messaging user to transfer to another extension or external phone number once logged onto DMS VoiceMail. This feature is not applicable to Call Answering (VMUIF) subscribers.
- ***Custom operator revert*** - This feature allows a caller who has been connected to a user's mailbox to press "0" and connect to an operator, secretary, etc. Since users can customize this number from their own telephone sets, it is important to restrict the extensions/phone numbers they try to use as their revert DN.
- ***Outcalling*** *(Remote Notification and Delivery to Non-Users)* - Remote Notification allows a user to be notified at a remote phone or pager when a new message arrives in his or her mailbox. Users can define their own remote notification schedules and target DNs from their telephone sets. Delivery to Non-users allows a DMS VoiceMail user to compose and send a message to someone off-switch who is not a DMS VoiceMail user. (Note that Delivery to Non-users is not available to VMUIF Call Answering subscribers.)

- ***External call sender*** - This feature is applicable only if Meridian Networking is installed. This feature allows a DMS VoiceMail user to immediately call back the sender of a just listened to message by pressing “9”. (This only applies to messages that have been left during call answering sessions, not voice messages that have been composed and sent.)
- ***AMIS networking*** - This feature is only applicable if AMIS Networking is installed. This feature allows DMS VoiceMail users to compose and send messages to sites that have voice messaging systems other than DMS VoiceMail. You may want to restrict users from sending messages to certain places.

The following field is displayed in the Add and View/Modify a Thru-Dial Definition screens:

- ***Thru-Dialers*** *(a type of voice menu application)* - Thru-dialers can be stand-alone applications or part of a larger voice menu application. When an external caller is connected to a thru-dialer, he or she is asked to enter the extension of the person he or she wants to talk to. If off-switch numbers are not restricted, an external caller is able to place local or long-distance calls from your switch. (Thru-Dialers can only be created if Voice Messaging is installed on the system.)

Securing your system involves defining restriction and permission codes which are then applied to specific features. Any dialing code can be entered as a restriction or permission code. A dialing code can be an extension number (on the switch) or any telephone number prefix that is used for dialing out of the switch (such as “9” for local calls or “91” for long distance calls, “6” for ESN calls, etc.).

You can define up to four different classes of service. This is done in the Voice Security Options screen. Each class can contain up to 10 restriction codes and 10 permission codes. The default classes are: On Switch, Local, Long Distance 1, and Long Distance 2. You could use the OnSwitch class to allow only extensions on the switch and restrict all local and long distance calls. You could use Local to allow local calls but restrict all long distance calls. You could use Long Distance 1 to restrict all long distance calls. You could use Long Distance 2 to restrict most local calls, but allow for calls to specific area codes. These are only suggestions and the manner in which you define you restriction classes will depend on your situation.

Once the restriction classes are defined, you can apply a particular restriction/permission class to each of the above features. For example, you could apply Local to extension dialing, but apply On Switch to the custom operator revert. Note that you can only apply one restriction class to a feature.

Restriction codes: 0 1 2 3 4 5 6 7 8 9
Permission codes: none

The Local class is assigned to all of the above features by default.

This essentially means that all possible dialing codes are restricted for mailbox thru-dial, custom operator revert, outcalling, external call sender and outbound AMIS messages. For these features to work at all, you must modify the restriction/permission codes to allow for certain dialing codes. This precaution forces you to customize these dialing codes to suit your particular situation.

To define and apply restriction/permission codes:

1 Define restriction/permission codes in the Voice Security Options screen.
2 Ensure that all user models have the appropriate restriction/permission codes set for the following features: mailbox thru-dial (extension dialing), custom revert, external call sender, AMIS networking, and outcalling features.

   See the section “User Models” in the “User Administration” chapter.
3 Restriction/permission codes can be customized for each user that you add to the system. See “Adding local voice users” in the “User Administration” chapter.
4 Apply restriction/permission codes to any Thru-Dialers (the voice menu application, not mailbox thru-dial) that you create. See the section “Thru-Dialers” in the “Voice Administration” chapter.

Security for system administrators

The administration terminal is password protected. When DMS VoiceMail is first installed, there is a default password. The first time you log on to the DMS VoiceMail administration terminal, you are forced to change this default password. You are recommended to change this password on a regular basis to maximize system security. Passwords can be between 1 and 16 characters in length. However, it is recommended that the password be no less that 7 digits in length. The longer the password, the less probable it is that someone will manage to guess it correctly.

Every time the administrator changes the logon password, a SEER (system event and error report) is generated, indicating this change. A SEER is also generated every time there is a failed logon attempt. This allows you to be aware of any attempts to breech the system’s security.

See the chapter “Administrator logon and the main menu” for more information.

You should investigate system security and overall system status whenever any of the following occurs:

- The administration password no longer provides system access (because it has been changed or locked out due to too many invalid logon attempts);
- A SEER indicates that the administrator password was changed (without the administrator's knowledge); or
- A SEER indicates a failed administrative logon attempt.

Security for mailbox users

DMS VoiceMail provides several ways of protecting user mailboxes against unauthorized access. This is mainly accomplished through the use of mailbox passwords.

The following parameters are configured in the Voice Security Options screen as described in the “Voice Administration” chapter.

Password change

The mailbox password is changeable by both the administrator and the mailbox user. It can be altered as often as desired. To compel mailbox users to change their passwords frequently, you can specify how often users are required to change their passwords. For example, you can require users to change their passwords every 30 days. The default is “0” meaning that users are not required to change their passwords.

This parameter is configured in the Voice Security Options screen and is called Maximum Days Permitted Between Password Changes.

Password length

Mailbox passwords can be between 4 and 16 digits in length. The greater the number of digits used in a password, the greater the security. You can specify the minimum password length in the Voice Security Options screen (the default is 4).

Invalid logon attempts

To guard against unauthorized access, you can have mailboxes automatically lock users out when a certain number of invalid mailbox logons have been attempted. There are actually two parameters which can be configured in the Voice Security Options screen: Maximum Invalid Logon Attempts Permitted per Session (the default is 3) and Maximum Invalid Logon Attempts Permitted per Mailbox (the default is 9). When this maximum limit is reached, the user's mailbox is disabled and must be re-enabled by the administrator.

External logon

This feature is “Enabled” by default, thus allowing users to log on to their mailboxes from phones that are external to the switch. If security is of the highest priority you can have this feature disabled. This option must be ordered from a Northern Telecom sales representative and implemented by a field technician. Once external logon is disabled on a system, it cannot be re-enabled.

Remote access restriction

DMS VoiceMail is configured with an internal modem which can be turned on at the local site in order to enable remote access to the system.

> ***Note:*** To ensure the security of your system, do not enable remote access unless absolutely necessary.

If the modem is currently disabled, press **Ctrl-w m** to enable it. This allows access to the system from a remote terminal. While remote access is enabled, you cannot access the system from the local terminal.

To disable the modem (and remote access), press **Ctrl-w m**. This re-enables local access. In this state, the system cannot be accessed from a remote terminal.

See the section “Using a remote terminal” in the chapter “Administrator logon and the main menu” for more information.

Setting up the system

Once the DMS/SL-100 has been provisioned, you are ready to set up the DMS VoiceMail system. Before beginning, review the site-specific and user-specific information you have prepared using the forms in the *Planning and Engineering Guide* (NTP 297-7001-100).

This chapter outlines general procedures and provides page references to sections that provide detailed information about the various aspects of configuration. Read the appropriate sections before configuring the system.

To start your configuration of DMS VoiceMail, begin with the basic setup procedure described in Procedure 3-1 to:

- check that DMS VoiceMail is operational,
- change the system administration password,
- configure general operating characteristics of the system and,
- back up the system with the new configuration information.

After you have completed the basic setup, refer to the other procedures in this chapter when you are ready to configure specific features, some of which are optional and may not be installed on your system. The other procedures in this chapter include:

- Configuring outcalling features
- Creating a voice menu application
- Configuring the AMIS Networking Service
- Configuring Meridian Networking
- Maintaining and servicing your DMS VoiceMail system

Basic setup procedures

Before carrying out any of the following steps, ensure that DMS VoiceMail has been properly provisioned on the DMS/SL-100. See the *Planning and Engineering Guide* (NTP 297-7001-100) and the *Translations Guide* (NTP 297-7001-310).

The following steps are common to all DMS VoiceMail installations, and are necessary for your system's operation.

Procedure 3-1xxx9z
Setting up the system

Step 1. Check the system status.	
From the logon screen, press the [System Status] softkey to ensure that the DMS VoiceMail system is operational.	See page 4-4.
Step 2. Change the system administrator password.	
Log on to the administration terminal with the default password (adminpwd). You are prompted to change the password the first time you try to log on.	See page 4-7.
Step 3. Check the hardware configuration.	
Check the node configuration and data port configuration.	
From the Main Menu, select Hardware Administration, Node Configuration.	See page 9-3.
If the configuration is incorrect, log on to the TOOLS menu, access HW_Modify and correct the configuration.	See "Hardware Modification" in Appendix A, "System Administration Tools".
Check the data port configuration to verify the correct assignment of data devices, especially parameters such as the baud rate and parity for the administration console.	
From the Main Menu, select Hardware Administration, Data Port Configuration.	See page 9-14.
If the configuration is incorrect, log on to the TOOLS menu, access HW_Modify and correct the configuration.	See "Hardware Modification" in Appendix A.
Step 4. Configure General Options.	
From the Main Menu select General Administration, General Options. Do the following: Enter the System Name (this is the name that will appear on reports). Enable SEER printing (if you have a printer attached to the administration terminal). If this is not enabled, SEERs will be displayed on your administration terminal. Assign an Attendant DN. If DMS VoiceMail is unable to handle a call, it is reverted to this number. Each user can have a custom revert DN. The system number you enter here is used as the default when adding users. (An attendant DN cannot be configured if Call Answering is installed on the system.)	See page 7-4.

Step 5. Check the Channel Allocation Table.	
The Channel Allocation Table should be properly configured after software installation. However, you may want to check it to ensure that the appropriate Primary DN and Channel DN have been assigned to the T1 channels in your system. From the Main Menu select Voice Administration, Voice System Configuration/Voice Menu Applications Administration, Channel Allocation Table.	See page 8-44.
Step 6. Add voice service DNs.	
From the Main Menu select Voice Administration, Voice System Configuration/Voice Menu Applications Administration, Voice Services-DN Table. You must add a DN for each voice service that will be directly dialable to users and external callers (such as the Voice Messaging DN, the Express Messaging DN, and any voice menu applications.)	See page 8-50.
Step 7. Customize voice messaging options.	
From the Main Menu select Voice Administration, Voice Messaging Options. If Voice Messaging is installed on the system: Record a custom call answering greeting. Configure the broadcast mailbox number. Enable users to record their own personal verifications (if desired). Set the maximum message length. Set the maximum amount of time that user's read messages are kept before being deleted by the system. If your system is multilingual, select the default language and secondary default language. If Call Answering is installed on the system: Record a customized VMUIF introductory tutorial (this is played when users log on for the first time). Configure the lockout revert DN. Configure the maximum message length. Configure the maximum amount of time that user's read messages are kept before being deleted by the system.	See page 8-5.

Step 8. Customize voice security options.	
From the Main Menu, select Voice Administration, Voice Security Options. Define restriction and permission codes. **THIS IS A VERY IMPORTANT STEP! These codes are applied to features like mailbox thru-dial, express messaging thru-dial, custom revert DN, extension dialing, call sender, thru-dialers (the voice menu application), and outgoing AMIS messages.** **All of these features are initially restricted. This means that none of these features will work until you modify the restriction/permission codes to allow certain external phone numbers or internal extension numbers to be dialed.** Set the maximum number of invalid logon attempts that a user is allowed to make before being locked out of his or her mailbox. If Voice Messaging is installed, change the default parameters that affect user passwords (such as the number of days allowed between password changes, the minimum password length, etc.).	See page 8-16.
Step 9. Configure the operational measurement options.	
This step does not need to be done right away. You may choose to use the default settings at first. Once the system has been in use for a while you can decide if the level of detail is adequate.	
From the Main Menu, select Operational Measurements, Operational Measurement Options.	See page 11-15.
Step 10. Customize user models.	
User models are templates that can be used when adding a large number of users with identical needs. Up to 15 models can be defined, three of which are pre-defined. User models should reflect the needs of certain types of users, such as managers, secretaries, and sales people.	
From the Main Menu, select User Administration, View/Modify User Models.	See page 6-84.
Step 11. Add users to the system.	
Read the section on “Planning how to add new users to the system” before beginning.	See page 6-5.
From the Main Menu, select User Administration, Add New User.	
Add local voice users. These are users that are on your system and that have a mailbox. Start with one user model (such as Secretary) and add all of the users that belong to this user type.	See page 6-12.
Add Directory Entry Users for people who want to be accessible by name dialing, but who do not need a mailbox. (Only if Voice Messaging, MMUI, is installed.)	See page 6-8.
If Networking is installed, users at remote sites can be added as Remote Voice Users (the reasons for doing this are explained in the “User Administration” chapter).	See page 6-52.

Step 12. Create distribution lists.	
This step does not have to be part of the initial configuration. If you know which lists you will need to create at this time, you may do so so that they will be ready for you to use. If you are unsure at this point, these can be created at any time. From the Main Menu, select User Administration, Distribution Lists.	See page 6-94.
Step 13. Back up the system.	
Once the system configuration has been customized, back up the new data onto tape to ensure its safety.	See page 7-8.

Setting up the Outcalling feature

The Outcalling feature refers to two functions. The first allows DMS VoiceMail users to be notified of new messages at remote phone or pager numbers and is known as Remote Notification (RN). The other feature, Delivery to Non-Users (DNU) allows users to compose and deliver messages to non-users of DMS VoiceMail. You may not have to change any of the parameters if you find that the default values are adequate. However, you should look over the default configuration to ensure that your organization's specific requirements are met.

> ***Note:*** Delivery to Non-Users is not available if Call Answering (VMUIF) is enabled.

Certain parameters are defined for the system as a whole. For details see "Outcalling administration" in the "Voice Administration" chapter. Other parameters are configured on a per-user basis in the Add or View/Modify Local Voice User screen. These screens are described in the "User Administration" chapter.

The following steps allow you to set up and customize the outcalling parameters on your system (if the Outcalling feature is installed on your system).

Procedure 3-2xxx
Configuring outcalling features (no new users are being added to the system)

Step 1. Configure outcalling options.	
From the Main Menu, select Voice Administration Administration, Outcalling Administration, Outcalling Options. Specify the following in this screen: Enable or disable audit trail data collection. If enabled, specify long should it be stored on disk. Specify the maximum number of remote notification retry repeats. Enter the data that will display on numeric pagers (this is the number that users must call back to retrieve messages). For Voice Messaging systems only: Specify how many times a delivery to non-user attempt should be retried if the remote phone or pager is busy, unanswered or answered without login. Specify how long DMS VoiceMail should wait before retrying a delivery to a non-user if the last attempt was unsuccessful due to the remote phone being busy, unanswered, or answered without login. Specify the time period during which delivery to non-users is allowed (there are geographical restrictions to electronic message delivery which must be respected). Enter the delivery to non-user addressing prefix. (This number is entered by users when composing messages to non-users and is a way of informing the system that the address is that of a non-user.) Specify how many times a message should be played to a non-user. Indicate whether or not non-users need to provide confirmation before receiving a message.	See page 8-121.
Step 2. Enable RN and DNU for each user.	
Both remote notification and delivery to non-users are disabled by default. These features must be enabled for each user that requires them.	
From the Main Menu, select User Administration, View/Modify User, [Local Voice User], [Change Defaults], [Outcalling Fields]. Set the *Delivery to Non-Users Capability* and *Remote Notification Capability* fields to “Yes”.	See page 6-46.

Step 3. Specify user-specific outcalling parameters.	
From the Main Menu, select User Administration, View/Modify User, [Local Voice User], [Change Defaults], [Outcalling Fields]. For DNU, specify the following: Indicate whether or not non-users need to provide confirmation before receiving a message. Specify which restriction and permission codes apply. ***Note:*** If Call Answering (VMUIF) is installed, you cannot enable DNU for users. For RN, specify the following: For Voice Messaging (MMUI) users, enable the keypad interface if you want the user to be able to modify remote notification schedules from his or her telephone set. Specify whether the user wants to be notified of any messages or just urgent ones. Specify which restriction and permission codes apply. Specify how many times remote notification should be retried if the remote phone or pager is busy, unanswered or answered without login. Specify how long DMS VoiceMail should wait before retrying remote notification if the last attempt was unsuccessful due to the remote phone being busy, unanswered, or answered without login. Define business days. This information is used to create remote notification schedules.	See page 6-46.
Step 4. Create remote notification schedules for users.	
For Voice Messaging systems, users can create their own remote notification schedules (if you enable the Keypad Interface in the user profile). If Call Answering is installed, you will have to create remote notification schedules for any users that want this feature. From the Main Menu select, User Administration, View/Modify User, [Local Voice User], [Change Defaults], [Outcalling Fields].	 See page 6-49.

Procedure 3-3xxx
Configuring outcalling features (new users are being added to the system)

Step 1. Configure system-wide outcalling options.	
From the Main Menu, select Voice Administration Administration, Outcalling Administration, Outcalling Options. Specify the following in this screen: Enable or disable audit trail data collection. If enabled, specify how long the data should be stored on disk before being deleted. Enter the maximum number of remote notification retry repeats. Enter the data that will display on numeric pagers. (This is the number that users must call back to retrieve messages.) For Voice Messaging (MMUI) systems only: Specify how many times delivery to non-users should be retried if the remote phone or pager is busy, unanswered or answered without login. Specify how long DMS VoiceMail should wait before retrying a delivery to a non-user if the last attempt was unsuccessful due to the remote phone being busy, unanswered, or answered without login. Specify the time period during which delivery to non-users is allowed. (There are geographical restrictions to electronic message delivery which must be respected.) Enter the delivery to non-user addressing prefix. (This number is entered by users when composing messages to non-users and is a way of informing the system that the address is that of a non-user.) Specify the number of times that a message be played to a non-user. Specify whether or not non-users need to provide confirmation before receiving a message.	See page 8-121.
Step 2. Enable RN and DNU in your user models.	
Both remote notification and delivery to non-users are disabled by default. These features must be enabled for each user that requires them. Therefore, if you have not yet added users to the system and know that you want all or some of your users to have these capabilities, configure your user models first and enable RN and/or DNU. ***Note:*** DNU cannot be enabled if Call Answering (VMUIF) is installed on the system. From the Main Menu, select User Administration, View/Modify User Models, [Change Defaults] [Outcalling Fields]. Set the *Delivery to Non-Users Capability* and *Remote Notification Capability* fields to “Yes”.	See page 6-89.

Step 3. Specify user-specific outcalling parameters.	
This can either be done in the user models (if outcalling will be configured similarly for all or most of your users) or when you add a new user.	
From the Main Menu, select User Administration, View/Modify User Models, select the model you want to modify, press [View/Modify User Model], [Change Defaults], [Outcalling Fields], or	See page 6-89.
From the Main Menu, select User Administration, Add New User, [Local Voice User], [Change Defaults], [Outcalling Fields].	See page 6-46.
For DNU, specify the following:	
Specify whether or not non-users need to provide confirmation before receiving a message.	
Specify which restriction and permission codes apply.	
For RN, specify the following:	
For Voice Messaging (MMUI) users, enable the keypad interface if you want the user to be able to modify remote notification schedules from his or her telephone set.	
Specify whether the user wants to be notified of any messages or just urgent ones.	
Specify which restriction and permission codes apply.	
Specify the number of times that remote notification should be retried if the remote phone or pager is busy, unanswered or answered without login.	
Specify how long DMS VoiceMail should wait before retrying remote notification if the last attempt was unsuccessful due to the remote phone being busy, unanswered, or answered without login.	
Define business days. (This information is used to create remote notification schedules.)	
Step 4. Create remote notification schedules for users.	
For Voice Messaging systems, users can create their own remote notification schedules (if you enable the Keypad Interface in the user profile). If Call Answering is installed, you will have to create remote notification schedules for any users that want this feature.	
From the Main Menu select, User Administration, Add New User, [Local Voice User], [Change Defaults], [Outcalling Fields].	See page 6-49.

Setting up optional features

DMS Voice Mail provides a number of optional features including Voice Menus, Voice Forms, AMIS Networking, and Meridian Networking. Meridian Networking is a proprietary Northern Telecom network feature and is available only on CPE systems.

Voice Menu Applications

The voice menus feature is optional and may not have been installed on your system. If Call Answering is installed on the system, voice menus cannot be installed.

The following steps allow you to create Voice Menu Applications and install them on your system.

Procedure 3-4xxx
Creating a voice menu application

Step 1. Design a voice menu application.	
Determine the purpose of the voice menu and the function(s) it is intended to serve. Plan the voice menu on paper first.	
Step 2. Define business hours and holidays.	
From the Main Menu, select Voice Administration, Voice Security Options. Define the Business Hours Default and Holidays. These definitions are used by any time-of-day controllers that you will create.	See page 8-20.
Step 3. Create a voice menu application.	
Add the voice menu application definition. From the Main Menu, select Voice Administration, Voice System Configuration/Voice Menu Applications Administration. From here you can add an Announcement Definition, Thru-Dialer Definition, Time-of-Day Control Definition or Voice Menu Definition.	 See page 8-62.
Step 4. Make the application accessible.	
Once you have created an application, you must make it accessible by:	
making it directly dialable (create a DN for it in the VSDN table); or	See page 8-50.
include it in a voice menu; or	See page 8-96.
include it in a time-of-day controller.	See page 8-85.

Voice Form Applications

Voice forms are an optional feature and may not be installed on your system. Voice forms are not available if Call Answering (VMUIF) is installed on the system.

Both the Voice Forms Implementation Guide and the section “Voice Form Definitions” in the “Voice Administration” chapter provide detailed procedures for configuring voice forms. Please refer to these documents.

AMIS Networking

AMIS networking is an optional feature and may not be installed on your system.

As explained in detail in the chapter “AMIS Networking”, you do not have to configure a DN specifically for AMIS networking because both voice menus and thru-dialers can accept incoming calls and pass them on to the appropriate AMIS agent. The only requirement is that the voice menu or thru-dialer have DID access. If the voice menu feature is not enabled, or if none of your voice menus/thru-dialers have DID access, you will have to configure a DN specifically for the AMIS service in the VSDN table.

Procedure 3-5xxx
Configuring the AMIS Networking service

Step 1. Configure AMIS networking information.	
From the Main Menu, select Network Administration, (AMIS Networking Administration if necessary), View/Modify AMIS Networking to access the View/Modify AMIS Networking Information screen. Configure the AMIS Compose prefix. Configure the Country Code and Area/City Code for the System Access Number. Configure the Country Code, Area/City Code, and Local Number for the System Access Number. Configure the permitted time periods for outgoing messages. Configure prefixes for public dialing, long distance dialing, and international dialing. The following parameters are configured with default values which may not have to be changed. Check the defaults. If they are not suitable, change them. Configure the wakeup interval, batch threshold, and networking call maximum. Configure initiation time for economy class messages. Configure the holding times for standard class and urgent class messages. Configure the stale times for economy, standard and urgent messages. Specify whether or not co-resident office codes are required.	
Step 2. Enable AMIS networking for users and apply restriction/permission codes.	
If you haven’t added users yet, you can modify your user models so that AMIS is enabled for the new users you add. From the Main Menu, select User Administration, View/Modify User Models. Select the model you want to modify. Set the fields *Receive AMIS messages* and *Compose/Send AMIS messages* to “Yes”. (Note that Call Answering subscribers can only receive AMIS messages.) Choose the restriction/permission codes that are to apply to AMIS calls.	See page 6-89.
If you have already added users and wish to enable AMIS for them: From the Main Menu, select User Administration, View/Modify User, [Modify Local Voice User], enter the user’s mailbox number. Set the fields *Receive AMIS messages* and *Compose/Send AMIS messages* to “Yes”. (Note that Call Answering subscribers can only receive AMIS messages.) Choose the restriction/permission codes that are to apply to AMIS calls (for Voice Messaging Users only).	See page 6-20.

Step 3. Make the AMIS service available to incoming AMIS calls.	
Option 1: If any of your voice menus or thru-dialers have DID access (are directly dialable external to the switch), have one of these services accept incoming AMIS calls. From the Main Menu, select Voice Administration, Voice Services Profile. Set the field *Act on AMIS Initiation Tone* to “Yes”. Publish one of your externally dialable voice menu or thru-dialer numbers as your AMIS number. When an incoming AMIS networking call is received, it will be recognized as such and will be passed on to an AMIS agent.	See page 8-59.
Option 2: *If you don’t have any voice menus or thru-dialers with DID access, configure a DN for the AMIS service in the VSDN table.* *From the Main Menu, select Voice Administration, Voice System Configuration, Voice Services-DN Table, [Add].* Enter an Access DN, enter AN (AMIS networking) as the Service.	See page 8-53.

Meridian Networking

Meridian networking is an optional feature and may not be installed on your system.

Once networking hardware has been installed, configure the networking service in DMS VoiceMail using the following procedure. Note that the first step is usually carried out by the system administrator.

Procedure 3-6xxx
Configuring the Meridian Networking service

Step 1. Configure a line DN (or a UCD queue) on the DMS/SL-100.	
If this has not already been done, create a DN for the Networking service if it will be sharing the agents in the primary voice messaging queue, or a UCD queue if agents will be dedicated to it. If you create a UCD queue with dedicated agents, modify the Channel Allocation Table. For the agents that have been moved to the new service queue, you will have to change the Primary DN to the UCD DN of the new queue.	See the section “Configuring voice services on the DMS/SL-100” in the Voice Administration chapter.
Step 2. Plan networking parameters.	
Make sure you have a good picture of what your network looks like (the number of remote sites, the dialing plan implemented at each site, etc.) Plan out the network on paper first. To gather appropriate information, consult with administrators at other DMS VoiceMail network sites to ensure the validity of information as well as to ensure that there is no duplication of location prefixes.	See pages 13-1 to 13-12.
Step 3. Add the Networking service DN to the VSDN table.	
Enable the service by entering the directory number in the VSDN table.	See page 8-50.
Step 4. Define attributes for the local site.	
From the Main Menu select, Network Administration, Local Site Maintenance.	See page 13-18.
Step 5. Define attributes for remote sites.	
Each site in the Meridian network must be defined in your database as a remote site. From the Main Menu select, Network Administration, Remote Site Maintenance.	See page 13-22.
Step 6. Check remote sites.	
Check that all remote sites have been added by going to the List Remote Sites screen.	See page 13-22.
Step 7. Test the network.	
Check the network by doing a loopback networking verification test.	See page 13-38.
Step 8. Back up the network database.	
Once you have defined the local site and added remote sites to your system, back up this new data to ensure its safety.	See page 7-8.

Routine maintenance and service procedures

The following steps are carried out regularly to ensure efficient operation of your system and to anticipate future needs concerning system capacity and types of services offered to users.

Procedure 3-7xxx
Maintaining and servicing your DMS VoiceMail system

Step 1. Monitor DMS VoiceMail operation.	
Check the performance of your DMS VoiceMail system periodically to ensure that efficient use is made of the voice services provided on your system.	See the “Operational Measurements” chapter.
Step 2. Monitor DMS VoiceMail hardware.	
Check the operation of DMS VoiceMail hardware periodically, or when a problem is reported by the system.	See the “System Status and Maintenance” and “Hardware Administration” chapters.
Step 3. Modify user information.	
User information can change periodically, due to relocation, change in classification, or the addition of new equipment and services. Such changes need to be reflected in the user information.	See the sections “Find Users ”, “View/Modify User”, “Delete User” and “Modifying a Distribution List” in the “User Administration” chapter.
Step 4. Back up the system.	
When changes are made to your system, back up the new data to ensure its safety.	See “Volume Administration” in the “General Administration” chapter.

Administrator logon and the main menu

Once the DMS VoiceMail system has been installed and the software is loaded, you are ready to log on to the system to gain access to the system administration menus, the starting point for initial setup of the system and general administrative functions.

Administrative functions can be carried out from the main administrative console attached to your DMS VoiceMail system or from a remote terminal connected to the system through a modem. User administration can be carried out on one of up to three user administration terminals (UATs), if configured. (See Appendix A, “System Administration Tools”, for more information about configuring UATs.)

A remote administration configuration is shown in Figure 4-7. If your installation uses this feature for the purpose of support from service personnel, you must coordinate remote administration sessions. See “Using a remote terminal” later in this chapter.

The Logon screen

The Logon screen is displayed when the administrative terminal is idle. From this screen you can log on to the administration console to set up and maintain your system,carry out administrative tasks on a system-wide basis or on a per-user basis, configure various voice services, or use the softkeys on the Logon screen to view the system status or DSP port status screens, or silence any alarms.

One of two logon screens will be displayed, depending on the option that was selected during installation. If CO (Central Office) was selected, the screen shown in Figure 4-1 is displayed. If CPE (Customer Premise Equipment) was selected, the screen shown in Figure 4-2 is displayed.

Figure 4-1xxx
The DMS Voice Mail Logon screen for CO environments

```
                              Logon/Status                              MM8



                   ____________________________________________________________
                  /                                               _______________  /
                 /                                               /  CENTRAL     / /
                /                                               /   OFFICE     / /
               /  ###       ### VOICE                          /______________/ /
              /   ###      ###                                                  /
             /    ###     ###     ###        ###  MAIL                         /
            /     ###    ###     #####     #####                              /
           /      ###   ###     ######   ######                              /
          /       ###  ###     ### ### ### ###                              /
         /        ### ###     ###  #####  ###                              /
        /         ######     ###   ###   ###                              /
       /          #####     ###         ###                              /
      /           ####     ###         ###                              /
     /            ###     ###         ###                              /
    /             ##     ###         ###                              /
   /                                                                 /
  /_________________________________________________________________/

                                    Copyright  (c)  Northern Telecom, 1992
 ______________________________________________________________________________
 Select a softkey >

     Logon          |            |   System    |   Silence   |   DSP Port
                    |            |   Status    |   Alarm     |   Status
```

Note 1: When logging on at a secondary terminal to perform User Administration, only the [Logon] softkey is displayed.

Note 2: Sometimes when you power down your terminal and then power it back up, the screen is drawn incorrectly. Namely, instead of the line that appears near the bottom of the screen (above the softkeys), a row of “q”s appears instead. Should this ever happen, do the following in order to redraw the screen: Press Ctrl-w (a small window opens up). Type **if**. (You do not have to press <Return>. The “i” means initialize and the “f” means full screen.)

Figure 4-2xxx
The Meridian Mail Logon screen for CPE environments

```
                              Logon/Status                                  MM8

              MMM       MMMERIDIAN
            MMMMM     MMMMMM
           MMMMMM   MMMMMMM
          MMM MMM MMM  MMM     MMM       MMMAIL
         MMM  MMMMMM  MMM    MMMMM     MMMMM
        MMM   MMM    MMM    MMMMMM   MMMMMM
       MMM          MMM    MMM MMM MMM MMM
      MMM          MMM    MMM  MMMMM  MMM
     MMM          MMM    MMM   MMM   MMM
    MMM          MMM    MMM         MMM
   MMM          MMM    MMM         MMM
  MMM          MMM    MMM         MMM
 MMM          MMM    MMM         MMM
MMM          MMM    MMM         MMM

                                     Copyright  (c)  Northern Telecom, 1992
______________________________________________________________________________
Select a softkey >

   Logon                       System         Silence        DSP Port
                               Status         Alarm          Status
```

Note: When logging on at a secondary terminal to perform User Administration, only the [Logon] softkey is displayed.

System Status

The System Status screen (Figure 4-3) is displayed when you press the [System Status] softkey on the Logon screen. The System Status screen is a read-only screen that dynamically updates when the status of system, system nodes or DSP ports changes. If you have to courtesy down the system, you must access the System Status screen from the System Status and Maintenance menu. See “System Status” in the chapter “System Status and Maintenance”.

Figure 4-3xxx
System Status screen

```
                                System Status


System Status: InService     Alarm Status: Critical=Off Major=Off Minor=Off


Last Event: 60-00 PRM: All System Programs Started   5/31 14:03

                    DSP Port/Channel Status                StorageUsed
Node Node  Status Active Idle OutSv Faulty Pending Others Voice     Text

 1   MSP   InService
 2   MSP   InService
 3   SPN   Faulty      0   0      0      12      0       0     32%    40%
 4   SPN   OutOfService0   0      12     0       0       0     41%    6%
13   T1FN  InService
14   T1FN  InService


  Select a softkey >

     Exit
```

Procedure 4-1xxx
Viewing the system status

Starting point : The Logon screen.

1. Press the [System Status] softkey to view the status of your system.
2. Use [Exit] to return to the Logon screen.

DSP Port Status

The DSP Port Status screen (Figure 4-4) is displayed when you press the [DSP Port Status] softkey on the Logon screen. This screen is read-only. It is dynamically updated as the status of your DSP ports change. If you suspect that one of your ports is not functioning properly, check this screen. To enable or disable a DSP port or perform out of service diagnostics, you must access the Card and/or DSP Port Status screen from the System Status and Maintenance Menu. See “Card Status” and “DSP Port Status” in the “System Status and Maintenance” chapter.

The example shown in Figure 4-4 illustrates the status for each DSP port with varying numbers of ports per node. Each node can have up to 24 DSP ports.

Figure 4-4xxx
The DSP Port Status screen

```
                                   DSP Port Status

DSP Port Status

       Ports
Node   1  2  3  4  5  6  7  8  9  10  11  12  13  14  15  16  17  18  19  20  21  22  23 24

3
4   .  .  .  .  a  .  a  a
5   a  a  .  a  .  .  .  .  .   .   .   a
6   .  .  .  a  a  a  .  .  .   .   .   .   .   .   .   a
7   a  a  .  .  O  O  O  O  a   .   .   .   .   a   a   .   .   .   .   .   F  F   F  F


  a = Active/In use     . = Idle/      O = Out of Service     U = Unknown
  F = Faulty            P = Pending    space = Unequipped     R = NoResource
  L = Loading

Select a softkey >
    Exit
```

Procedure 4-2xxx
Viewing the DSP Port Status screen

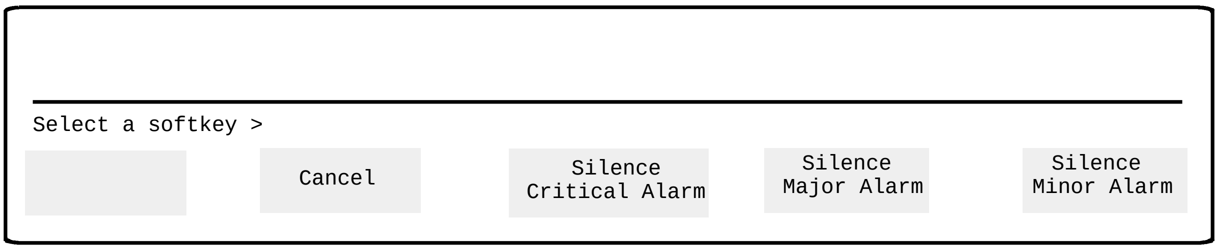
Starting point : The Logon screen.

1. Press the [DSP Port Status] softkey to view the status of system ports.
2. Use [Exit] to return to the Logon screen.

Silencing Alarms

When the system sounds an alarm, you may silence it using the [Silence Alarm] softkey on the Logon screen. When this softkey is pressed, the softkeys displayed in the following figure are displayed.

Figure 4-5xxx
The Silence Alarm softkeys screen



An alarm will sound if the corresponding severity level SEER is issued indicating that a problem exists. By using the appropriate softkey you can silence either critical, major, or minor alarms. The [Cancel] softkey causes the original set of softkeys to be displayed without silencing any alarms. Try to clear the problem as well or the alarm could be turned on again if you simply silence it. Alarms persist until you silence them (there is no timeout period after which they are turned off by the system.)

For more information on alarms, refer to the *Trouble locating and alarm clearing procedures* (NTP 297-7001-503) and the *Maintenance Messages* manual (NTP 297-7001-510).

Logging on as the System Administrator

When you press the [Logon] softkey you are prompted for a password. If you are logging on for the first time, use the default password **adminpwd**. When you log on for the first time with the default password, the system prompts you for a new password. The system does not allow you to log on until you have changed the default password.

Passwords can be up to 16 characters in length. However, it is recommended that the password be no less that 7 digits in length for added system security. The longer the password, the less probable it is that someone will manage to guess it correctly.

You should continue to change the logon password on a regular basis to ensure the security of your system. In the future, you will change the password from the General Administration menu.

Procedure 4-3xxx
Logging on as the System Administrator

Starting point : The Logon screen.

1 Press [Logon]. Enter the system administrator password and press <Return>. If the system has been down due to a power outage or some other problem, the system prompts you to enter the date and time. Enter the date and time in the format indicated, with leading zeroes, slashes, and colon (e.g., 31/01/89 09:35).

If an invalid password is entered, an error message appears. Try logging on again.

Note: *If you are logging in for the first time, you will be prompted to change the default password. To do so, enter a new password and press <Return>. You are prompted to re-enter the password for verification. Enter the password again and press <Return>. If you entered the password incorrectly the second time, you will have to enter the password again.*

The Main Menu is displayed (Figure 4-6). From the Main Menu you will specify the specific administrative task you want to perform, such as adding users and configuring their mailboxes, configuring voice services , backing up your system, checking your hardware configuration, reading operational measurement reports, and performing system maintenance. The various administrative tasks are described throughout the rest of this guide.

2 Use [Logoff] to return to the Logon screen.

Note: *An unsuccessful logon attempt is automatically recorded in the system log file. As a security precaution, after a third unsuccessful attempt to log on, the system forces a ten minute delay before a further logon attempt will be accepted. Only your Northern Telecom representative has the requisite privileges to gain access to the system during the lockout period.*

CAUTION
If you forget your password
If you have forgotten your password, you will have to reboot the system from the install tape. When the system boots from the tape, an item is presented which allows you to reset the password to the original default. Once this has been done, the install tape can be removed from the tape drive and the system will reboot from the disk. Once the system is up, use the default password to log on. You will be prompted to change it immediately. Use a memorable yet non-obvious password.

Logging on as User Administrator at a UAT

If the Multiple Administration Terminal (UAT) feature is configured, your DMS Voice Mail system can support up to four administration terminals (one main administration terminal and up to three secondary terminals.) Logging on to a secondary terminal gives you access to User Administration screens only.

The logon password is the same for all terminals (the main administration terminal and all secondary user administration terminals). The default password is **adminpwd**. You can only change this password at the main administration terminal. A password change is automatically carried over to the configured UATs.

If you log on to a secondary terminal with the default password, you will be prompted to enter a new password immediately. (The system will not allow you to log on until you have changed the default password.)

When you log on successfully, the User Administration menu is immediately displayed (the Main Menu that is displayed on the main administration terminal is not shown).

For information about configuring multiple administration terminals, see the section “Configure UATs” in Appendix A.

Procedure 4-4xxx
Logging on to a secondary terminal as the User Administrator

Starting Point : The Logon screen.

1. Press the [Logon] softkey.
2. Enter the password and press <Return>.

   If an invalid password is entered, an error message appears. Try logging on again.

   If the password is valid, the User Administration screen is displayed. See the “User Administration” chapter for information on user administration.

The Main Menu

> ***Note:*** If multiple administration terminals have been configured, the Main Menu is only displayed on the main administration terminal. Secondary terminals only display the User Administration Menu.

This menu is a routing menu from which you can select the type of administrative function you require. The [Exit] softkey is also displayed and should be used once you have completed your administrative activities.

> ***Note:*** For security and memory usage reasons, do not leave the administrative console unattended while you are logged on. Also, remember to log out at night. If you do not log out, critical audit and backup routines may not be able to run due to insufficient memory.

Figure 4-6xxx
The Main Menu

```
                       Main Menu

            1.  User Administration

            2.  General Administration

            3.  Voice Administration

            4.  Hardware Administration

            5.  System Status and Maintenance

            6.  Operational Measurements

        *   7.  Network Administration


 Select an item >

   Exit
```

```
* This option is displayed only if Meridian Networking and/or AMIS
  Networking is installed.
```

Procedure 4-5
Navigating the Main Menu

Starting point : The Main Menu.

1 Choose an item by entering its number and pressing <Return>.

The appropriate menu appears. See the following chapters for details:

"User Administration";

"General Administration";
"Voice Administration";
"Hardware Administration";
"System Status and Maintenance";
"Operational Measurements";
if installed, "Network Administration";

2 Carry out the required administrative functions, then return to the Main Menu; repeat step 1 to carry out additional administrative tasks, or proceed to step 3.

3 Use [Exit].

The Logon screen is redisplayed.

Resetting the system time

It is possible that the system time may be undefined, as may happen when a time signal is not provided by the switch to which DMS VoiceMail is connected or when a time signal is provided but the link to the switch is temporarily down. In both cases, the system automatically prompts you for the correct time. You cannot proceed with administrative functions unless the system date and time are defined.

You may be required to enter the time at the Logon screen, under unusual circumstances such as power outages. At other times, you can perform optional system time changes as desired. See “Changing the system time” in the chapter “General Administration”.

CAUTION
Setting the time ahead
If you set the time ahead by a number of days (if for example, the current time is incorrect or you are testing time of day controllers), all read messages that meet the Read Message Retension Value (set in the Add Local Voice User screen) will be deleted. For example, today is December 9th and the read message retention limit is 7 days. You set the time ahead by 72 hours. Any messages that are 4, 5 or 6 days old will be deleted before they are supposed to be according to the read message retention maximum.

Procedure 4-6
Resetting the system time

Starting point : Logon screen, system time incorrect or undefined after logon.

1 You are prompted for the correct time. Enter the date and time in the format indicated, with leading zeroes, slashes, and colon (e.g., 31/01/89 09:35).

The Main Menu is displayed.

2 Use [Cancel] if you choose not to set system time.

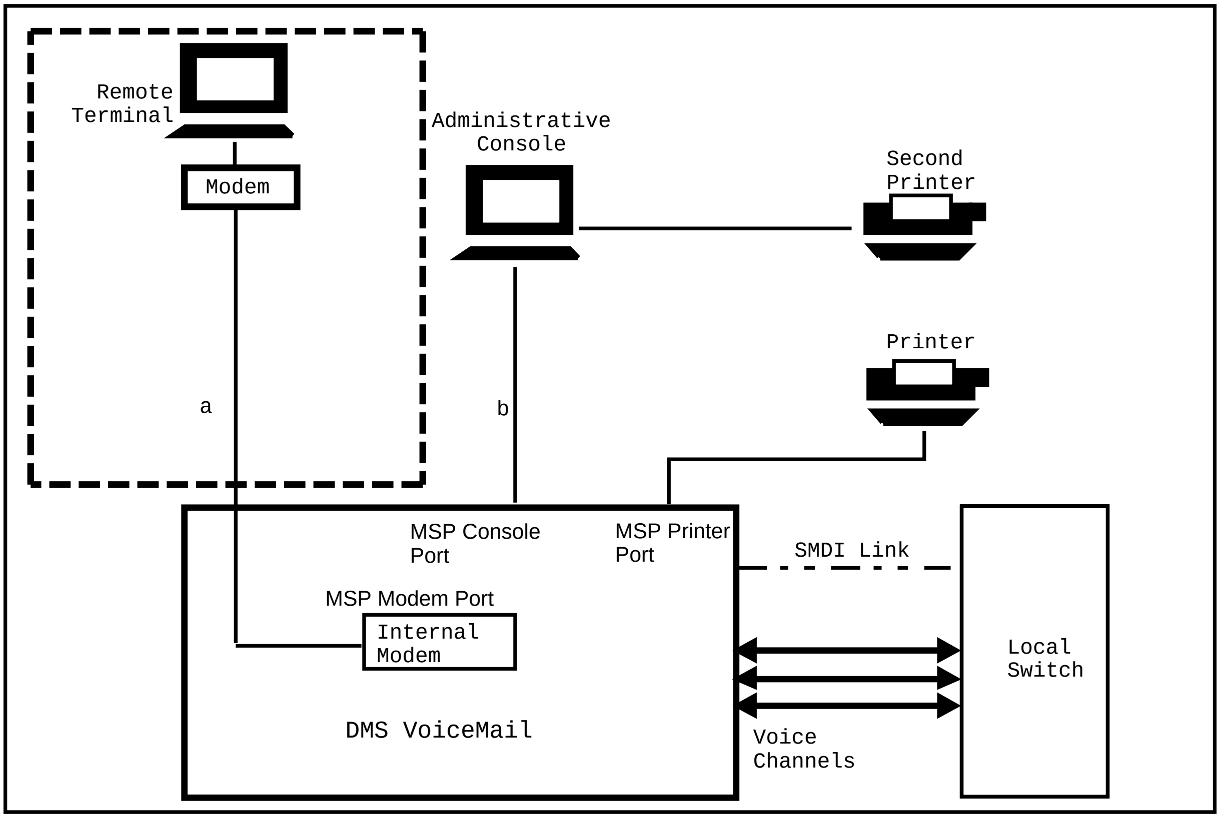
The password prompt is redisplayed.

You may wish to investigate the source of the time discrepancy; see DMS VoiceMail Trouble locating and alarm clearing Procedures (NTP 297-7001-503).

Using a remote terminal

If your installation has a remote terminal installed for service personnel, as shown in Figure 4-7 (Connection Option a), the remote access user can log on to the system to perform administrative functions once remote access has been enabled at the local terminal. While a remote logon is in effect, no administrative functions can be carried out from the local console. (When remote access is disabled, a remote user cannot log in to the system.) You should therefore schedule remote logins with the remote user for a time when you will not require access to the system.

Figure 4-7xxx
A typical remote administration configuration



“Appendix C: Remote Access” in the *System Application Software Installation and Modification Guide* (NTP 297-7001-504) provides information needed to set up a remote terminal and modem.

Procedure 4-7xxx
Enabling/disabling remote access

Starting point : The Logon screen, at the local system administration console.

1 To bring up the COBRAVT selection window, enter **Ctrl-w** (while holding down the <Ctrl> key, press <w>).

> ***Note:*** For help using COBRAVT, type a question mark (?). A help screen is displayed.

2 Type **m** (case does not matter).

3 Notify the user at the remote terminal.

The Logon screen appears at the remote console.

The remote user hits the <Break> key to gain control of the console.

The remote user enters the administration password to gain access to the system.

The administrative functions described in this manual are identical when viewed from the local or remote administrative terminal.

4 To disable remote access, repeat steps 1 and 2 at the local administration terminal.

Control is returned to the local console, and the Logon screen is redisplayed.

*You can terminate a remote logon by entering **Ctrl-w m** at the local console at any time during the remote log on.*

Note: *This may cause data loss if the remote administrator is in the process of changing system data and a save was not performed.*

On-line Help

As described earlier in the chapter “Understanding DMS VoiceMail administration”, on-line help is available for most of the menus and administration screens, including the Main Menu. The <Help> key on the keyboard can be used to display information on whatever screen you are working in. If you require help with a screen, press the <Help> key. The system will display explanations of all the fields on the menu or screen you are working in. When you are done, use the [Exit] softkey on the Help screen to return to the screen you are working in.

Making recordings

Guidelines for making voice recordings

Prompts used solely for administrative purposes can be recorded without much preparation other than deciding on the exact wording of the prompt. For voice menus or announcements played to the public or members of your organization, more formal preparation may be necessary. The following is a list of guidelines you may wish to use when recording prompts:

- Use a voice that is similar to the DMS VoiceMail prompts and consider using only one voice to avoid distracting callers by changes in pitch, tone, intonation, or accent. Choose a voice that suits your organization's image. Select the person who will read the text and print complete, definitive copies of the script. Audition a few candidates by recording their voices, then playing the recordings over the telephone line. Low-pitched voices are reproduced over telephone lines better than high-pitched ones.
- Record in quiet surroundings.
- Start recording immediately after the tone and stop the recording immediately after the last word. This prevents unnecessary pauses when system prompts and Personal Verification recordings are joined together.
- Do not hang up the phone while recording as this may produce clicks in the recording. Instead press **#** to stop recording.
- For applications that provide current information, it is perhaps best to have the person who knows the information monitor the prompts to ensure that the information is always up-to-date.
- When recording a Personal Verification for two or more people in your organization who have the same name (or very similar names), provide more information (their extension number or title, for example) to distinguish them.
- Record a few names for Personal Verification and listen to them before recording the remaining names. This ensures that the procedure is done correctly and the intonation is good. Test each of the following areas where Personal Verification applies:
  - call answering greeting (unless there is a personal greeting already recorded)

- message envelope playback
- address playback in the compose command
- name dialing, name addressing.

The call answering greeting and personal verifications

These greetings are used for identification purposes. One identifies your organization to external callers and the other identifies users during message composition.

> ***Note:*** These greetings only apply if Voice Messaging is installed on the system.

The call answering greeting

This greeting identifies your organization to external callers. The greeting typically consists of the spoken name of the organization. It is played when a user's mailbox is reached through call answering or express messaging. It is also played by the remote notification service during notification delivery.

This greeting is optional. If recorded, external callers hear this greeting before the user's personal greeting. If you do not record a custom greeting, no call answering greeting is played and callers simply hear the user's personal greeting when they reach a mailbox.

Because this greeting is used in a variety of situations, you will have to consider how to best word this greeting (or decide if you want to record a greeting at all). For example, during remote notification calls, the following prompt is played to MMUI users if no call answering greeting is recorded: *"Hello. DMS VoiceMail has received a message for ..."*. For users belonging to VMUIF customer groups, the prompt is *"Hello. Call Answering has received a message for ..."*.

When a custom call answering greeting exists, the following prompt is played: *"Hello. <Call Answering Greeting> has received a message for ..."*. If the call answering greeting is something like *"Hello. Thank you for calling Myelin Incorporated"*, the prompt will not sound right when used during remote notification. Consider the following when deciding whether or not to record a call answering greeting.

- If you do not record a call answering greeting, the organization's name will not be announced at the beginning of a call answering or express messaging greeting. When an external caller is connected to a user's mailbox, the caller will only hear the user's external greeting (or internal greeting, if an internal but not external greeting is recorded). If you feel that the user's personal greeting is sufficient, you may regard this greeting as unnecessary.

- If you record just the organization's name (*"The Myelin Corporation"*), the greeting that is played during call answering may sound too abrupt. However, the prompt that is played during remote notification will sound quite natural.
- A friendlier greeting (*"Thank you for calling The Myelin Corporation"*), is ideal for call answering scenarios, yet results in an awkward sounding prompt for remote notification (and express messaging since this feature is used by users residing on the same switch).

The personal verification

The personal verification is a recording of a user's first and last names (and extension, if desired). It is used to identify the owner of a mailbox. If no verification is recorded, the system plays a recording of the user's extension number. Since it is easier to determine if you have reached the correct person by hearing their name than hearing their extension number, it is highly recommended that a personal verification is recorded for all users with mailboxes.

> ***Note:*** Personal verifications can only be recorded for users if Voice Messaging is installed on the system. It is not intended for Call Answering (VMUIF) subscribers.

The personal verification can be recorded by you as you add each user to the system or by the users themselves. If you want users to record their own verifications, you will have to enable this feature in the Voice Messaging Options screen (see page 8-7). The field is called *User Changeable Personal Verification* and it is disabled by default.

The procedure for recording personal verifications at the administration terminal is described in the "User Administration" chapter. However, it is ideal to have users record their own personal verifications because the user's own voice is likely to be more recognizable to callers. The user's procedure for recording a name for personal verification is covered in the *DMS VoiceMail Quick Reference Guide*. If you prefer that users record their own personal verifications, ensure that they are informed of this feature and that they are instructed in the procedure.

Personal verifications are played in the following situations:

- During message composition, the personal verification is played after the mailbox number is entered to verify that the correct person is being addressed.
- Messages delivered to non-users (using the Delivery to Non-Users feature) include the personal verification. The recipient of the message will be more likely to listen to the message if they recognize who the message is from.

- When a user is called using the name dialing feature, the personal verification is played instead of spelling out the name to the caller.
- During remote notification the system will play the verification to identify for whom the message is intended.

  Note: You can also record verifications for users as you add them to the system. This is done using the [Voice] softkey on the User Administration menus. See the chapter “Making recordings” for more information.

Procedure 5-1
Recording call answering greetings and personal verifications

1 Log on to a DMS VoiceMail mailbox with administrator capabilities.

2 Follow 2a to record a call answering greeting or 2b to record Personal Verification recordings.

a. To record a call answering greeting, press **829** on the telephone keypad.

b. To record a Personal Verification for a user, press **89**, enter the user’s mailbox number and then press **#**.

3 Choose step 3a to replace an existing call answering greeting or Personal Verification, or 3b to add a new greeting or verification.

a. Press **76** to delete the old greeting. Proceed to 3b.

b. Press **5** to start recording.

If a previous recording exists, the added recording will be appended to the existing message.

4 Wait for the tone and say the custom call answering greeting or Personal Verification (name of user).

5 Press **#** to stop recording. (Do not hang up the phone during recording as this may produce a click sound.)

6 To check the recording, press **2** (play).

7 When recording is finished, press **83** to end the voice messaging session, then hang up.

Broadcast Messages

There may be times that you will need to send a message to all users. A message that is sent to all users is known as a *broadcast message.* A special mailbox number (the broadcast mailbox number) is defined in the Voice Messaging Options screen (see the chapter “Voice Administration”). When composing a broadcast message, you simply specify the broadcast mailbox number and all users in the system will receive it. Note that any user who knows the broadcast mailbox number and has access to a mailbox with administrator capabilities can also send broadcast messages.

> ***Note:*** It is recommended that you refrain from sending broadcast messages during busy hours.

It is a good idea to record a personal verification for the broadcast mailbox (before you record and send any broadcast messages as described in Procedure 5-2). This verification is played to users when they receive the message. You can either identify who the message is from (i.e., the system administrator) or that the message is a broadcast message so that each recipient knows that all users have received the message. This verification is recorded from the Voice Messaging Options screen using the [Voice] softkey. See the section “Voice Messaging Options” in the “Voice Administration” chapter for details.

Procedure 5-2xxx
Sending broadcast messages

> ***Note:*** *If you have not recorded a personal verification for the broadcast mailbox, do so from the Voice Messaging Options screen before beginning this procedure.*

1 Log on to a mailbox with administrator capabilities.

2 Press **75**, enter the broadcast mailbox number, and press **#**.

3 Press **#** again to end the list.

4 Press **5** to start recording.

5 Wait for the tone and say the message to be broadcast.

6 Press **#** to stop recording.

7 To check the recording, press **2** (play).

8 To send the broadcast message, press **79**.

9 When the message is sent, press **83** to end the voice messaging session, then hang up.

Voice Prompt maintenance

If you delegate the task of maintaining recordings used in voice menu applications (voice menus, thru-dialers, and announcements), ensure that your delegates are trained in using the Voice Prompt Maintenance service. You can also use this service when you must re-record prompts frequently. The service allows you to review and modify voice prompts through a DTMF telephone rather than the administrative console.

Though prompts cannot be deleted through the Voice Prompt Maintenance Service, recording a new prompt automatically overwrites any previous prompt. You cannot update a voice recording through Voice Prompt Maintenance Service while it is being updated through the Voice Menu Applications Administration function. Callers also hear the old version of the menu, thru-dialer or announcement while it is being updated.

Voice menu applications (voice menus, announcements and thru-dialers) contain recorded data or prompts of one kind or another. An announcement contains just one recorded prompt which is played back to callers. A voice menu contains an introductory greeting as well as a prompt which specifies the actions which a user can take by pressing keys on the telephone keypad. Thru-dialers also contain an introductory greeting. Prompts can be recorded by the administrator from the administration terminal, or by using the Voice Prompt Maintenance Service.

If voice menu applications are to be maintained by administrative delegates, an Update Password must defined for the application (see “Voice Menu Applications Administration” in the “Voice Administration” chapter). If no Update Password is assigned, the menu or announcement will not be accessible through the Voice Prompt Maintenance Service; it can only be updated through Voice Menu Applications Administration.

Procedure 5-3xxx
Playing and recording announcements and thru-dialer greetings

1 Dial the Voice Prompt Maintenance Service DN.

The system prompts you for an ID.

2 Enter the required Announcement ID or Thru-dialer ID and press #.

The system prompts you for the Update Password.

3 Enter the Update Password and press #.

You are prompted to use Play or Record (Use Play to hear the entire prompt from start to finish).

4 Play the announcement or greeting, or update it and save the new announcement.

Record overwrites what was there before.

5 To return to the ID prompt, enter a number sign.

You can update another announcement or thru-dialer greeting by going to step 2.

Procedure 5-4xxx
Updating voice menu prompts

1 Dial the Voice Prompt Maintenance Service DN.

The system prompts you for an ID.

2 Enter the required Voice Menu ID and press #.

The system prompts you for the Update Password.

3 Enter the Update Password and press #.

4 The system plays a menu with four choices:

a. Update Greeting prompt

b. Update Menu Choices prompt

c. Update No Response prompt

d. Update Other Menu prompts

5 Select the required function.

If you select a, b, or c you are prompted to play the prompt if it exists.

If you select d, you are prompted for the number of the prompt. This number is the number of the key a caller using the menu must press to hear the prompt. Enter the number.

6 Play or record the prompt.

If you selected d after playing, recording, or updating the prompt, enter a number sign (#) to go back to where you can enter the (key) number of another prompt.

7 To return to the ID prompt, enter a number sign.

You can now work on another menu by going to step 2.

Remote Activation

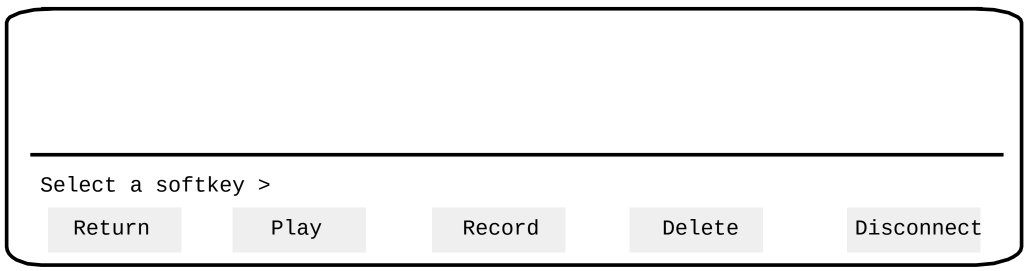
Remote Activation allows administrators or delegates to associate a DN with a different voice menu application (voice menu, announcement, thru-dialer, time-of-day controller, or voice form, if installed) from off-site, using a standard DTMF telephone set. Use this feature in the event that access to the switch location is disrupted (during a storm for example). For more information see “Using remote activation” in the “Voice Administration” chapter.

Making recordings using the [Voice] softkey

The [Voice] softkey is displayed on some administration screens. It can be used to record personal verifications and prompts for voice menu applications and voice forms. If the environment around your terminal is noisy, you may prefer to use a phone that is in a quieter location to dial into the Voice Prompt Maintenance Service to record voice menu prompts, or to record the call answering greeting or personal verifications. When the [Voice] softkey is pressed, a new set of softkeys is displayed. See Figure 5-1.

> ***Note:*** A telephone set is required to make recordings. Ensure that a phone set is available near the administration terminal where you are working.

Figure 5-1
Recording softkeys



Procedure 5-5
Using the recording softkeys

1 Press the [Voice] softkey.

You are prompted for an extension number.

2 Enter the extension number of the phone set you are going to use to make the recording.

The phone will ring when you finish entering the extension.

3 Pick up the telephone handset.

4 To record, go to step 4a. To listen to the existing recording, go to step 4b. To delete the existing recording, go to step 4c. To return to the original set of softkeys, go to step 4d.

a. Press the [Record] softkey. At the sound of the beep begin speaking into the handset.

When you pressed the [Record] softkey, a new [Stop] softkey appeared in its place. Press the [Stop] softkey to stop recording.

b. Press the [Play] softkey.

If a recording has already been made, it is played over the phone.

c. Press the [Delete] softkey.

If a recording has been recorded, it is deleted. A prompt is displayed advising you that the recording was deleted.

d. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the recording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

Playing a recording

The voice recording can be played using the [Play] softkey.

Procedure 5-6xxx
Playing a voice recording

Starting point The current screen, Voice softkeys displayed.

1 Use [Play].

If there is no current recording, a message is displayed on the console.

If a recording is available, it is played, and the [Stop] softkey is displayed;

2 Use [Stop] at any time to stop the playback.

The Voice Recording softkeys are redisplayed.

Recording a new message

The voice recording can be recorded using the [Record] softkey. This overwrites any existing recording.

Procedure 5-7xxx
Recording a voice recording

Starting point The current screen, Voice softkeys displayed.

1 Use [Record].

A message is displayed on the console requesting you to make the recording, and a beep can be heard in the telephone receiver.

The [Stop] softkey is displayed.

2 Say the text of the recording and use [Stop] when you are done.

The Voice Recording softkeys are redisplayed.

The recording will be stopped automatically if you exceed the Maximum Prompt Size or the Record Timeout set in the Voice Service Profile screen.

If a recording existed before, it is overwritten.

Deleting a recording

The recording can be deleted using the [Delete] softkey.

Procedure 5-8xxx
Deleting a voice recording

Starting point The current screen, Voice softkeys displayed.

1 Use [Delete].

A message is displayed on the console requesting you to confirm the deletion; the softkeys [OK to Delete] and [Cancel] are displayed.

2 Choose 2a to delete the recording, or 2b to cancel.

a. Use [OK to Delete].

The recording is deleted.

The Voice Recording softkeys are redisplayed.

b. Use [Cancel].

The Voice Recording softkeys are redisplayed; the recording is not deleted.

User Administration

Categories of users

User administration primarily involves adding users to the system, and once added, maintaining the existing user profiles. When you add a new user to the system, you must specify the user type. There are three categories of users as described below:

- ***Directory Entry Users*** can only be added to the system if the Voice Messaging feature (MMUI) is enabled. These are typically Centrex business users who are registered in the DMS VoiceMail directory but who do not have a mailbox. As a result, they do not have access to voice messaging functions. They can, however, be referenced by such features as name dialing and automated attendant functions such as voice menus (if these are installed on your system).
- ***Local Voice Users*** have DNs on the local switch. If Voice Messaging (MMUI) is enabled, each local voice user has a mailbox with call answering capability. This means that if the user is away from his or her phone (or on the phone), callers are connected to their personal mailbox in which they can leave a voice message. Furthermore, these users have access to voice messaging functions (i.e., they can compose and send messages to other users and non-users).

  If Call Answering (VMUIF) is enabled, each local voice user has a mailbox from which messages can be retrieved. However, they only have access to call answering functions and do not have the ability to compose and send messages.

- ***Remote Voice Users*** are subscribers on other DMS VoiceMail systems who have access to your system through the Meridian Networking service (if installed). (Meridian Networking is an optional feature for CPE systems only.) Not all voice users at remote sites need to be added to your system as remote voice users. You may only want to do this for those users who most frequently call your site. When a user from a remote site sends a message to a user at the local site, the personal verification is not played unless the user is defined as a remote voice user in your system. (When a user is not defined as a remote voice user, that user's mailbox number is played instead of the personal verification which makes it harder for local voice users to identify the sender of a message.)

The User Administration screens

The User Administration screens provide the necessary facilities to add, modify and delete directory entry users, local voice users and remote users. A number of the fields related to the Outcalling feature (Remote Notification and Delivery to Non-Users) are set on a per-user basis in these User Administration screens.

The Find facility, also available under User Administration, simplifies the process of locating existing users for the purpose of modifying or deleting them.

The appearance of the user administration screens will vary depending on which of the following features is installed: Voice Messaging (full-featured voice messaging) or Simplified Call Answering (call answering only without voice messaging capability). Many of the fields in the user administration screens are specific to one of these features. The screens depicted in this chapter will therefore show a number of fields that may not apply to you. Wherever a field is dependent on a particular feature, this will be pointed out.

When you add users to the system, the user profile is based on a pre-existing user model. The models serve as templates to simplify the process of adding new users to the system. When you add a new user, the parameters default to the settings contained in the model on which the user profile is based. In this manner you do not have to explicitly configure every parameter for each user. If certain users require some parameters to be configured differently than the norm, you need only change those specific parameters. Fifteen such user models are provided for you. See the section “User Models” later in this chapter for more information.

Multiple administration terminals

DMS VoiceMail supports up to four administration terminals (one main administration terminal for system administration and up to three secondary terminals for user administration only). If your system has multiple administration terminals, only the first administrator who logs on to perform user administration on a particular user can modify or delete that user. Screens will be read-only for other administrators who then log on to the same user profile.

For information about configuring multiple administration terminals, see the section “Configure UATs” in Appendix A.

The User Administration Menu

The User Administration screen (Figure 6-1) provides six functions with which to manipulate user-related information.

Figure 6-1xxx
User Administration screen

```
                    User Administration


   1 Add New User

   2 View/Modify User

   3 Delete User

   4 Find Users

   5 View/Modify User Models

   6 Distribution Lists


 ------------------------------------------------------------
 Select an item >

 Exit
```

Procedure 6-1xxx
Navigating the User Administration screen

Starting point : Main Menu, <1> entered.

1 The User Administration screen appears (Figure 6-1).

2 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears. See the following sections for details:

<1> "Adding new users"
<2> "View/Modify User"
<3> "Delete User"
<4> "Find Users"
<5> "View/Modify User Models"
<6> "Distribution Lists"

3 Use [Exit] to return to the Main Menu (or the Logon screen if you are logged onto a secondary terminal).

Planning how to add new users to the system

Before you begin to add any users to the system, you should do some planning first. Ask yourself the following questions:

1 How are you going to distribute users over volumes to ensure that some of your volumes don't become full while others remain empty?

2 If you are adding large numbers of users in a short time period (i.e., in a 24-hour period), there are special considerations.

Distributing users over volumes

DMS VoiceMail systems can have from one to eight voice nodes, each of which contains a hard disk drive for data storage. The hard disk drives are partitioned into volumes. Volumes are storage areas for system and user related information. The volumes are already set up when your system is installed.

Users are not automatically distributed over volumes. When you add users, you must specify the volume to which the user is to be added. The default is volume "203". If you do not change this value in the Add a Local Voice User screen, all users will be added to volume 203 thus filling it up and leaving other volumes empty. Table 7-2 in the "General Administration" chapter, specifies the maximum amount of storage available on each volume for the various DMS VoiceMail configurations.

Be careful of how you assign users to volumes. Putting certain types of users who share the same usage pattern (especially those who use the system heavily) on the same volume increases the probability that too many channels will try to access the same disk at one time. For example, all secretaries are added to the same volume (volume 203). They all come in at 9:00 a.m. and log on immediately. Suddenly a large number of channels are trying to access the disk. This situation is not desirable. It is therefore recommended that you distribute users across volumes randomly in such a manner that does not result in correlations in access patterns among the users on a volume.

Before adding users to the system, survey your users to estimate average usage in terms of number of messages and length of each message. Compare this with the capacity of the available disk volumes and the minutes of storage you wish to assign to users, and estimate the number of users each volume can accommodate. Randomly assign users on different disks to distribute traffic evenly to the disk drives. Ideally, each user volume should have an equal number of users. For example, to randomly select users, choose the volume based on the first letter of the user's surname.

Note: The maximum voice storage for each mailbox is equal to the mailbox size plus the maximum message length.

Information on disk usage can be obtained through the Disk Usage report generated by the system administrator (see "Traffic Reports" in the "Operational Measurements" chapter). A listing of disk volumes can be obtained by displaying the Volume Administration screen . For information about volume names and how information is distributed on the volumes, see "Volume numbers and distribution".

If a volume becomes full and you need to move users to another volume, you can do so using the Move User utility. This utility is available under the Tools menu. To move a user you must know the user's mailbox number. For more information, see Appendix A, "System Administration Tools".

Adding large numbers of users

It is not recommended that you add a large number of users (600 or more) in a short period of time. (A short period of time here means a 24-hour period between two nightly audits. These audits take place between 2:30 a.m. and 5:00 a.m.). When you add such a large number of users, the organization directory which stores user profile information, can become unbalanced and perform less efficiently. The nightly audit rebalances the directory. If you must add a large number of users between audits, consider the following factors:

1. Ensure that the number of users to be added is within the engineering guidelines for the system. Specifically, DMS VoiceMail is engineered for up to 5,000 users per voice node.
2. If Voice Messaging (MMUI) is installed on your system, add users in *reverse alphabetical order*. Otherwise performance will gradually degrade after approximately 600 users have been added (the system gets slower and slower). This degradation in performance will be corrected when the next nightly audit occurs.

   If Call Answering (VMUIF) is installed, you can add users in any order.
3. Distribute users across volumes as evenly as possible. See the section "Distributing users over volumes" earlier in this chapter.
4. Do not add more than 2000 users to the same exchange. Otherwise, the system will become unbalanced. The next nightly audit will rebalance the system. For example, if the exchanges 763, 766 and 769 exist on your switch, do not add more than 2000 users to any of them within a 24-hour period.
5. For Voice Messaging users, be careful of how you fill in the Department field. Avoid broad categories which will place more than 100 users in a single department.

Adding new users

When you select the Add New User option from the User Administration screen, a new set of softkeys (shown in Figure 6-2) is displayed. They allow you to specify whether you want to add a directory entry user, a local voice user, or a remote voice user. Other than the new softkeys, the screen is identical to the User Administration screen.

Figure 6-2xxx
Add New User softkeys

```
                    User Administration

      1  Add New User
      2  View/Modify User
      3  Delete User
      4  Find Users
      5  View/Modify User Models
      6  Distribution Lists

   ______________________________________________

   Select an item >

   Exit      Cancel    Add Directory   Add Local    Add Remote
                       Entry User *    Voice User   Voice User**
```

```
* This softkey does not apply if Call Answering (VMUIF) is enabled.
**This softkey is only displayed if Meridian Networking is installed
   (for CPE systems only).
```

Procedure 6-2xxx
Adding a new user

Starting point: User Administration screen, <1> entered.

The following softkeys appear: [Add Directory Entry User], [Add Local Voice User], [Add Remote Voice User], and [Cancel].

1 Choose step 1a to add a directory entry user, 1b to add a local voice user, 1c to add a remote voice user, or 1d to return to the User Administration screen.

a. Select [Add Directory Entry User].

See the next section, “Adding directory entry users”, for details.

b. Select [Add Local Voice User].

See “Adding local voice users” later in this chapter for details.

c. Select [Add Remote Voice User].

See “Adding remote voice users” later in this chapter for details.

d. Select [Cancel].

The User Administration screen and the [Exit] softkey are redisplayed.

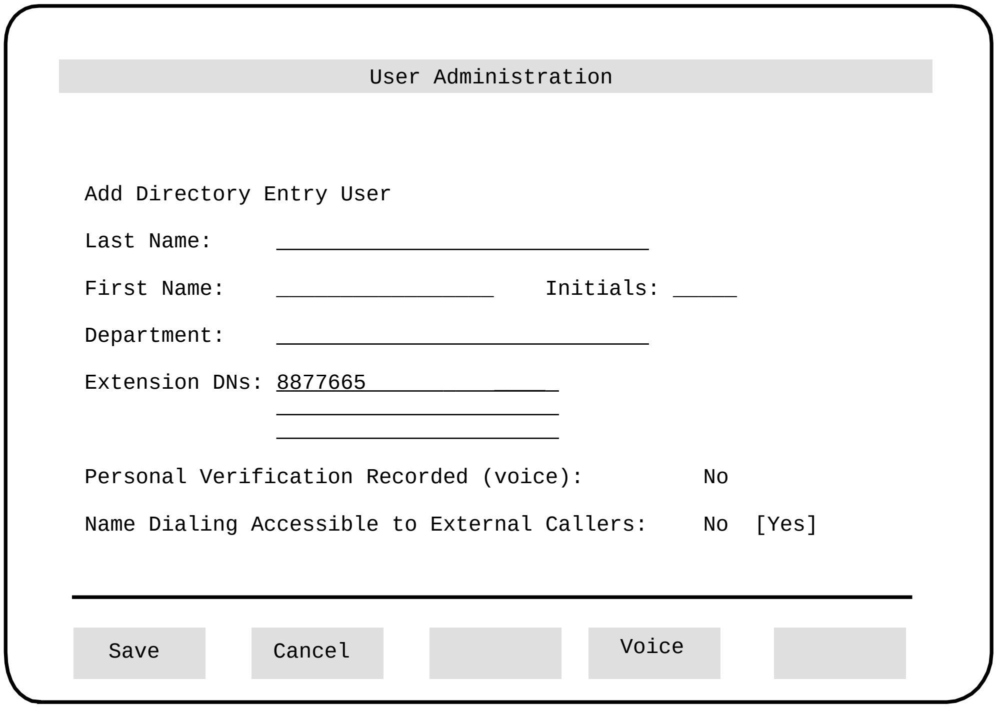
Adding directory entry users

Directory entry users do not have voice mailboxes associated with their extensions. This is useful, for example, when a telephone is used by a number of different people. You do not necessarily wish to create a mailbox for this type of phone, but you may wish to associate the names of the people who use the phone with the extension. Then other DMS VoiceMail users can dial the phone using thru-dial features such as Name Dialing.

> ***Note:*** Users cannot be added to the system as directory entry users if Call Answering (VMUIF) is installed on the system.

When you press the [Add Directory Entry User] softkey on the Add New User screen, a prompt appears asking you to enter the extension number (DN) to be added. When you enter the extension number, the Add Directory Entry User screen (Figure 6-3) appears. The screen contains fields that identify each directory entry user and associate the users with primary and optional secondary and tertiary extension numbers. Primary extension numbers are not unique; several users can share the same extension.

Figure 6-3xxx
Add Directory Entry User screen



The Directory Entry Users screen contains the following fields:

- ***Last Name*** - The last name of the new directory entry user, up to 41 characters in length. This field is mandatory. This field accepts any characters with the exception of the restricted characters “+”, “_”, and “?”. However, you should limit yourself to alphanumeric characters for name dialing and name addressing to work properly. This field is blank by default. Be sure to fill it in and ensure correct spelling because the Name Dialing and Name Addressing features use this information.
- ***First Name*** - The first name of the new directory entry user. You can enter up to 21 characters, including the space and hyphen (-). The default is blank. Ensure correct spelling because the Name Dialing and Name Addressing features use this information.
- ***Initials*** - The initials of the directory entry user. This field can hold up to 5 alphanumeric characters. This field is for display only and can be used by the administrator to distinguish users with identical first and last names. These initials, however, cannot be used in name dialing.

  Note: If you do not enter any initials, the system will automatically fill in this field with the first initial of the user’s first name.

- ***Department*** - The department to which the user belongs. You may enter up to 31 characters. The characters "+", "?" and "_" are restricted. It is recommended that you use alphanumeric characters only and avoid using special characters altogether (even though some are accepted by this field) for the reasons mentioned in the *Last Name* field. When adding the first user to the system, this field will be blank by default. For subsequent users, this field defaults to the department entered for the last user added.

  You can retrieve users on the basis of department when using the Find Users function (described later in this chapter).With Find Users however, only the first ten characters of the department are displayed. Therefore try to assign unique identifiers for each department. For example, if you have the departments Marketing Sales and Marketing Advertising, you should enter them as Sales Marketing and Advertising Marketing.

- ***Extension DNs*** - The user's extension number or numbers. A user's DN can be up to 30 digits in length. A user can be associated with three possible extensions. The first field is the user's primary extension number and is mandatory and defaults to the number entered to access the Add User screen.

  If you will only be entering a Primary DN in a CO environment, enter the 7-digit directory number. If you will also be adding a Secondary DN, enter the 7-digit DN as the secondary DN. The third field for the tertiary DN is optional.

  Note: On CO systems for which the SMDI link is set to 10-digit messaging, enter the full 10-digit DN (including the area code).

  In a CPE environment, the DN is typically four or five digits in length.

- ***Personal Verification Recorded (Voice)*** - If a personal verification has been recorded for this user, this field displays "Yes". "No" indicates that no verification is currently recorded. The setting in this field changes when the [Voice] softkey is used to record a verification (or when a user records their own verification from their telephone set). The personal verification is used in address lists, during call answering sessions and when name dialing is used.
- ***Name Dialing Accessible to External Callers*** **-** When this field is set to "Yes", outside callers who are required to dial an extension DN to reach a DMS VoiceMail user (when reaching a voice menu for example), can enter the user's name instead of the extension. This may not be desirable for all users as a caller who is connected to your system through a voice menu can get through to any extension as long as they know the person's name. You may therefore want to set this field to "No" for those users who have their phone calls screened by a secretary. This field defaults to "Yes".

Procedure 6-3xxx
Adding a directory entry user

Starting point : User Administration screen, [Add Directory Entry User] entered.

1 The [Cancel] softkey appears and you are prompted to enter an extension number.

2 Go to step 2a to return, or to 2b to proceed.

a. Use [Cancel].

The Add New User softkeys (Directory Entry, Local and Remote Voice User) are re-displayed; see "Adding new users" earlier in this chapter for details.

b. Enter the extension number and press <Return>.

The Add Directory Entry User screen appears with the cursor positioned at the Last Name field (Figure 6-3).

3 Enter the *Last Name*, *First Name*, *Department*, and *Extension Numbers* of the new user.

4 Set *Name Dialing Accessible to External Callers*, if necessary.

5 Use [Voice] to record a Personal Verification recording.

a. Enter the extension number of the phone you will be using to record.

A new set of softkeys is displayed.

b. Press the [Record] softkey.

c. At the sound of the beep, speak the user's name into the telephone handset.

d. Press the [Stop] softkey to stop recording.

e. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the re-cording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

See the section, "Recording personal verifications using the [Voice] softkey" on page 6-56 for more information about the recording softkeys.

6 Go to step 6a to save the new user, or 6b to cancel the addition.

a. Use [Save].

The system saves the user and prompts for another extension number. To add another user, enter the extension and press <Return>. Then go to step 2b. Go to step 6b to exit.

b. Use [Cancel].

Any new user information that entered is discarded. The Add New User softkeys are re- displayed; see "Adding new users" for details.

Adding local voice users

Most of the users that you add to the system will be local voice users. They are added to the system from the Add Local Voice User screen. Although each user profile is based on a particular user model, the fields in this screen are modifiable so that you can customize each user profile in order to meet the user's requirements and needs.

Before adding local voice users you should:

1. Determine the capacity of your disk volumes.
2. Survey users to determine the types of user models that will be necessary and to estimate the average system usage of each type of user.
3. Create user models to reflect the results of your survey.
4. Decide on a method of randomly assigning users to volumes.

When adding a large number of users, the best approach is to add them according to user model. For example, add all Secretaries, then Executives, etc. (keeping in mind to randomize the volume to which users are added).

User passwords

If Voice Messaging is installed on the system, each user must have a password. When you add a new user, the system assigns a default password (the user's mailbox number). This password can be changed by the administrator or by the user at the telephone set.

If Call Answering is installed, the system does not assign a password when a mailbox is added. However, you (or the subscriber) can define a password once the mailbox has been created. No password implies access from the "home phone" only.

> ***Note:*** A password must be created if a call answering subscriber wants to use the remote notification feature. A password is also necessary if the subscriber requires the capability to log on to his or her mailbox from a telephone other than his or her "home phone". You can create a password at the administration terminal using the [Change User Password] softkey in the Add (or Modify) Local Voice User screen. Alternatively, you can inform subscribers that they will have to create a password using their telephone set in order to use remote notification and to log on to their mailbox remotely.

Add Local Voice User screen

When you press the [Add Local Voice User] softkey in the User Administration menu, you are prompted to enter the mailbox number to be added. When the mailbox number has been entered, the Add Local Voice User screen (Figure 6-4) appears.

When you add the first new user to the system, the first *Extension DN* field defaults to the mailbox number you entered to gain access to this screen. When you add subsequent users, you need only fill in the following fields: *Mailbox Number*, *Last Name*, and *First Name*. *Extension Number* defaults to the value in the *Mailbox Number* field. All the other fields, including *User Model*, default to the values chosen for the previous user. This allows you to quickly add several new users who share the same basic information.

Procedure 6-4xxx
Accessing the Add Local Voice User screen

Starting point The Main Menu

1 Select User Administration.

2 Select Add New User.

3 Press the [Local Voice User] softkey.

You are prompted for an extension.

4 Enter the user’s DN followed by <Return>.

This number will be used in the Add Local Voice User screen to fill in the Mailbox Number and primary Extension DN fields.

If DMS VoiceMail is connected to an SL-100, this should be the number that is dialable by other local users, typically a 4-digit extension number. On the DMS-100, enter the user’s 7-digit directory number.

The Add Local Voice User screen is displayed.

Figure 6-4xxx
Add Local Voice User screen

```
                         User Administration

    Add Local Voice User

    Mailbox Number:  8765432____     Volume ID: 203

 *  User Model:       [Executive]    Secretary   Standard

    Last Name:      ______________________________________

    First Name:     __________________      Initials: ________

**  Department:     _______________________

    Extension DNs:  8765432 _____________
                    ____________________
                    ____________________

    Revert DN:      0___________________

    Personal Verification Recorded (Voice):   No

**  Name Dialing Accessible to External Callers: No  [Yes]

 ________________________________________________________________

   Save      Cancel     Change       Voice      Change User
                        Defaults                 Password
```

```
* If Call Answering (VMUIF) is installed, the User Models are DTMF,
  Deluxe and Dial_Pulse. The Revert DN default is blank.
**These fields are only displayed if Voice Messaging (MMUI) is
  enabled.
```

The following fields are displayed:

- ***Mailbox Number*** -This field is automatically filled in with the DN you entered to access this screen, although it can be changed from within this screen. In an office or centrex environment, this is the number that is dialable by other local users (typically a 4-digit DN). If this is a Call Answering subscriber, this is the 7-digit directory number. This field is mandatory. If it is not filled in, you will not be able to save the user profile.

The mailbox number can be up to eight digits in length and range between 11 and 99999999. The number should not conflict with any of the following: the broadcast mailbox number, the Name Dialing prefix (see “Voice Messaging Options” in “Voice Administration”), Delivery to Non-user dialing prefixes (see “Outcalling Administration” in “Voice Administration”), system distribution list numbers, other mailbox numbers, or the AMIS prefix. The initial setting is the extension you specified after pressing the [Add Local Voice User] softkey.

Note: People that are not in the office much (such as salespeople) may not have their own telephone set. You still can, however, configure a mailbox for these people so that they can collect and listen to messages. All that you need is an available DN on the switch that is not associated with a physical telephone set.

- ***Volume ID*** - This field specifies the hard disk volume to which the user is assigned. All users must be assigned to a volume. The default is “203”. For subsequent new voice users, this field defaults to the value set for the previous new user. For more information see “Distributing users over volumes” earlier in this chapter.
- ***User Model*** - This field indicates the user model upon which the new user profile is based. The three pre-named user models, as well as any models that you name in the View/Modify User Models screen, are displayed for selection. For the first user you add, the field defaults to the first user model in the list. For subsequent new voice users, the field defaults to the model set for the previous new user.
- ***Last Name*** - The last name of the new directory entry user, up to 41 characters in length. This field accepts any characters with the exception of the restricted characters “+”, “_”, and “?”. However, you should limit yourself to alphanumeric characters. If you use any control characters or special characters, name dialing and name addressing may not work properly. This field is blank by default. Be sure to fill it in and ensure correct spelling because the name dialing and name addressing features use this information.

  Important: If you must change a user’s last name once the mailbox has been added and in use, do not modify this field. Instead, make sure the user has listened to all of his or her messages, delete the mailbox and re-add it with the new last name. DMS VoiceMail uses the user’s last name to keep track of users, mailboxes and messages. Modifying the *Last Name* field can cause inconsistencies.
- ***First Name*** - The first name of the new local voice user. You can enter up to 21 characters, including the space and hyphen (-). However, you should limit yourself to alphanumeric characters for the reasons mentioned in the *Last Name* field. Ensure correct spelling because the Name Dialing and Name Addressing features use this information.

- ***Initials -*** The initials of the local voice user. This field can hold up to 5 alphanumeric characters. This field is for display only and can be used by the administrator to distinguish users with identical first and last names. These initials, however, cannot be used in name dialing.

  Note: If you do not enter any initials, the system will automatically fill in this field with the first initial of the user's first name.

- ***Department*** - The department to which the user belongs. You may enter up to 31 characters. The characters "+", "?" and "_" are restricted. It is recommended that you use alphanumeric characters only and avoid using special characters altogether (even though some are accepted by this field) for the reasons mentioned in the *Last Name* field. When adding the first user to the system, this field will be blank by default. For subsequent users, this field defaults to the department entered for the last user added.

  Note: This field does not apply if Call Answering (VMUIF) is enabled.

  You can retrieve users on the basis of department when using the Find Users function (described later in this chapter).With Find Users however, only the first ten characters of the department are displayed. Therefore try to assign unique identifiers for the each department. For example, if you have the departments Marketing Sales and Marketing Advertising, you should enter them as Sales Marketing and Advertising Marketing.

- ***Extension DNs -*** The user's extension number or numbers. A user can have up to three extension DNs defined in his or her user profile (a primary, secondary and tertiary DN). This means that a caller can dial any of these numbers and still reach the user's mailbox. A DN can be up to 30 digits in length.

  The first field is for the primary DN and is mandatory. You cannot save the user profile if this field is blank. It is automatically filled in with the DN you entered to access this screen and is therefore the same as the mailbox number.

  In a CPE environment, DNs are typically four or five digits in length.

  In a CO environment, DNs are typically seven digits in length.

  Note for CO systems: If the mailbox number and primary DN are not the user's 7-digit directory DN, you must enter the 7-digit directory DN as the secondary DN. However, if the SMDI link is set to 10-digit messaging, enter the full 10-digit DN (including the area code). If the 7-digit (or 10-digit) directory DN is the primary DN, the secondary DN is optional. The tertiary DN is optional.

- ***Revert DN*** - This is the number to which calls are passed when:
  - a caller presses “0” during a call answering session, or
  - when a user waits more than 2 seconds to enter “#” after dialing 0 in order to place a call while in his mailbox (known as mailbox thru-dial).

  If Voice Messaging is installed

  Users can also configure their own Revert DN through their telephone set. This is covered in the *DMS Mail Voice Messaging User Guide.*

  If this field is filled in, the user will have to include a statement in his or her external and internal greetings to inform callers that they can press the revert DN (usually “0”) if they want to be connected to a secretary or cellular phone.

  In an office or centrex environment, calls are normally reverted to back-up people such as secretaries or receptionists. The default value is “0”.

  If you are adding the first new user and you do not enter a DN, the system will use the System Attendant DN entered in the General Options screen (see “General Options” in “General Administration”). Once you begin adding users to the system, this field will default to the DN entered for the previously added user. The revert DN may be up to 30 digits in length and can begin with 0 (zero).

  If Call Answering is installed

  This field is blank by default. You can, however, enter a DN in this field if the subscriber requests this capability. A small business may ask for this feature so that calls can be reverted to a secretary. Call Answering subscribers may ask for this feature if they want callers to be able to try them at another number, such as that of a cellular phone. (Note that for Call Answering subscribers this DN only applies to call answering sessions because subscribers do not have mailbox thru-dial capabilities.)

- ***Personal Verification Recorded (Voice***) - The spoken name of the user can be recorded by the administrator using the [Voice] softkey or by the user at the telephone. When a verification is recorded, this field is updated to show “Yes”. Otherwise, it will show “No”. See the chapter “Making recordings” for more information about the personal verification and when it is used.

  Note: Personal verifications can only be recorded for users if Voice Messaging is installed on the system. It is not intended for Call Answering (VMUIF) subscribers. Although the administrator can still record a personal verification from this screen, it will never be used since Call Answering subscribers do not have access to the features that use the verification.

- ***Name Dialing Accessible to External Callers*** **-** This field is displayed only if the MMUI Interface is used and thus would not apply to Call Answering subscribers. When this field is set to “Yes”, outside callers who are required to dial an extension DN to reach a DMS VoiceMail user (when reaching a voice menu for example), can enter the user’s name instead of the extension. This may not be desirable for all users as a caller who is connected to your system through a voice menu can get through to any extension as long as they know the person’s name. You may therefore want to set this field to “No” for those users who have their phone calls screened by a secretary. This field defaults to “Yes”.

Procedure 6-5xxx
Adding a local voice user

Starting point : User Administration screen, [Add Local Voice User] entered.

1 The [Cancel] softkey appears, and you are prompted to enter a mailbox number.

2 Go to step 2a to return, or 2b to proceed.

a. Use [Cancel].

The Add New User softkeys are displayed; see “Adding new users” earlier in this chapter for details.

b. Enter the mailbox number and press <Return>.

The Add Local Voice User screen appears with the cursor positioned at the User Model field (Figure 6-4).

3 Select the desired *User Model*. Use the <Return> or <Tab> key to move through the list of available choices.

4 Enter the *Last Name, First Name, Initials, Department, Extension Number(s),* and *Revert DN* of the new user.

5 Use [Change Defaults] to display additional fields of the Add Local Voice User screen; see “Changing defaults for local voice users” later in this chapter for details.

6 Press [Voice] to make a Personal Verification recording.

A Personal Verification recording should be created for all Voice Messaging users. Personal verifications are not required for Call Answering subscribers.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Record] softkey.

c. At the sound of the beep, speak the user’s name into the telephone handset.

d. Press the [Stop] softkey to stop recording.

e. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the recording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

See the section, “Recording personal verifications using the [Voice] softkey” on page 6-56 for more information about the recording softkeys.

Note: *Users can record their own personal verifications if you enable this feature in Voice Messaging Options (see “Voice Messaging Options” in the “Voice Administration” chapter).*

7 Use [Change User Password] to set the user’s password.

You are prompted to enter the new password, then to re-enter the new password to verify it. The passwords are not displayed on the screen.

User passwords must be numeric and up to 16 digits long. By default, the initial password for a new user is the same as the user’s mailbox number. For Call Answering subscribers, there is no initial password.

8 Go to step 8a to save the new user, or 8b to cancel the addition.

a. Use [Save].

The system saves the new user and prompts for another local voice user’s mailbox number; go to step 2 to add another user, or to 8b to leave the menu.

b. Use [Cancel].

New user information is discarded. The Add New User softkeys are displayed; see “Adding new users” earlier in this chapter for details.

Changing defaults for local voice users

When you select the [Change Defaults] softkey on the Add Local Voice User screen, a full local voice user screen, displaying a detailed list of user-specific parameters appears (Figure 6-5). Some of the things you can do from this screen are: enable or disable name dialing, give the user’s mailbox administrative abilities, change the read message retention, allow or disallow the user to keep copies of sent messages, and set restriction and permission codes for features such as extension dialing, custom revert, external call sender, and AMIS (if installed).

The fields in this full voice user screen default to the settings stored in the user model used to create the user. Changes made to these default settings affect only the specific user, not the user model.

From this full screen additional outcalling parameters can be accessed by pressing the [Outcalling Fields] softkey. The outcalling fields are in the sections that follow this one.

Figure 6-5xxx
Add Local Voice User (Change Defaults) for the MMUI interface

```
                         User Administration

      Add Local Voice User

      Mailbox Number:     8765432        Volume ID:  203

      Storage Limit (minutes): 20        Storage Used:  12

      Last Name:     _________________________________________

      First Name:    ____________________________   Initials: ________

      Department:    ______________________________

      Extension DNs: 8765432 ____________________
                     __________________________
                     __________________________

      Revert DN:     0___________________

      Message Waiting Indication Option: None [Any]  Urgent

*     Message Waiting Indication DN:  ________________________________

*     Message Waiting Link name:      ________________________________

      Personal Verification Recorded (Voice): No
                                                          MORE BELOW
```

*   These fields are displayed only if MWI is “Any” or “Urgent”.

Figure continued on next page

Figure 6-5 (continued)
Add Local Voice User (Change Defaults) for the MMUI interface

```
                    User Administration                MORE ABOVE

   Add Local Voice User

   Name Dialing Accessible to External       No  [Yes]
   Callers:

   Administrator Capability:                 [No] Yes

   Logon Status:                             Disabled  [Enabled]

   Billing Class:                            2

   Read Message Retention (days):            10
   ("0" implies that read messages
   are retained until the user
   deletes them manually.)
   (Subject to the organization's
   maximum retention of 7 days)

   Retain Copy of Sent Messages:            No  [Yes]

   Auto Logon:                              [No] Yes

   Delayed Prompts:                         No  [Yes]

                                                               MORE BELOW
```

```
                    User Administration                MORE ABOVE

    Add Local Voice User

    Auto Play:                           [No] Yes

    Callers Notified of Busy Line:       No  [Yes]

  * Dual Language Prompting:             Disabled  [Enabled]

  * Preferred Language:                  [AmericanEnglish]
                                         EuropeanEnglish
                                         Mandarin

    Custom Revert                        None  On_Switch [Local] Long_distance_1
    restriction/permission codes:        Long_distance_2

    Extension Dialing                    None  On_Switch [Local] Long_distance_1
    restriction/permission codes:        Long_distance_2

 ** External Call Sender                 None  On_Switch [Local] Long_distance_1
    restriction/permission codes:        Long_distance_2

*** Receive AMIS messages:               [No]    Yes

*** Compose/send AMIS messages:          [No]    Yes

*** AMIS restriction/permission codes:   None  On_Switch [Local]   Long_distance_1
                                         Long_distance_2

  Save        Cancel       Outcalling       Voice        Change
                             Fields                      Password
```

```
*  This field is only displayed if the system is multi-lingual.
** This field is only displayed if Meridian Networking (for CPE only) is installed.
***These fields are only displayed if AMIS networking is installed.
```

When you press the [Change Defaults] softkey, the fields in the first Add a Local Voice User screen are displayed again. They are filled in with the values that you entered before pressing the [Change Defaults] softkey. These fields have already been described in the previous section “Adding local voice users” and will not be described here.

The following fields are displayed if the Voice Messaging feature (MMUI) is installed on the system. Some of these fields are prefilled with default settings. These defaults are based on the user model upon which the user profile is based. User Models are discussed on page 6-87.

- ***Storage Limit (minutes)*** - The maximum amount of storage available to the user. You may enter a value from 1 to 360 (minutes). This field defaults to the value set in the user model.

  If a user surpasses this limit his calls are not cut off. The user hears a message indicating that his mailbox is full and he is restricted in what he can do. For example, he can only read and delete messages and is not allowed to record a personal greeting, compose, send or forward messages. Once the user has deleted some of his messages, he won’t be able to reply to messages until he has logged off DMS VoiceMail and logged back on.

- ***Storage Used*** - This read-only field indicates how many minutes of voice messages are currently stored for the current user. This value is rounded up to the nearest minute. Before deleting a user, check this field to make sure that there are no voice messages in the mailbox.

  Note: A user may inform you that he or she has received the mailbox full warning, but that the mailbox is definitely not full. For example, the user is certain that there are only two short messages in the mailbox. A prematurely full mailbox is caused by an unexpected system reboot that leaves inconsistencies between the volume server and what is actually in the mailbox. This problem will be fixed automatically during the scheduled nightly audit. However, if an unexpected reboot happens at a busy traffic time, you can log on at the Tools level and select the menu item “Audit all volumes”. This will update the real mailbox storage information that is stored on disk and prevent prematurely full mailboxes. See the *System Administration Tools Guide* (NTP 555-7001-305) for more information about this tool.

- ***Message Waiting Indication Option*** - The chosen setting determines the type of messages that will cause a message waiting indication (a flashing light or an interrupted dial tone) on the user's telephone set. Set this field to "Any" to notify the user of all new messages, "Urgent" to notify the user of only those messages tagged as urgent, or "None" if the user is not to be notified at all (if, for example, the mailbox does not have a telephone set associated with it). The field defaults to the value set in the *User Model*. The following two fields (*Message Waiting Indication DN* and *Message Waiting Link Name*) are not displayed if "None" is selected.
- ***Message Waiting Indication DN*** - This field must be filled in if the *Message Waiting Indication Options* field is set to "Any" or "Urgent". This DN specifies the telephone extension at which message waiting is activated when a new message is put in the user's mailbox. This field defaults to the user's mailbox number. On CO systems, set this field to the user's 7-digit directory number as it is configured on the switch.

  Note: If the SMDI link is configured for 10-digit messaging, enter the user's 10-digit directory number (this DN includes the area code).

  This field should be set to "None" for users that don't have physical telephone sets, but does have a mailbox. For example, a salesperson may only rarely be at the office and does not have a phone as a result, but still requires a number for callers to leave messages.
- ***Message Waiting Link Name*** - This field is displayed only if *Message Waiting Indication Option* is set to "Any" or "Urgent". It specifies the link on which the message waiting indication is sent for this user.

  This field is intended for systems with the Multi-SMDI (indicated as "SMDI" in the General Options screen) feature so that you can distribute users over all available links. If you have only one SMDI link, this field defaults to the link name entered in the hardware database and cannot be changed from this screen.

  If you do have multiple SMDI links, do not put all users on the same link. Instead, distribute users (as evenly as possible) across all available links. This field defaults to the first link name defined in the hardware database. You can cycle through the other link names while in this screen. To display the next link name, delete the current link name and press <Tab> or <Return>. Repeat this procedure until you have cycled through all of the link names (you may want to write them down as you go). You can also view the the link names in the View Data Port screen (SMDI), as described in the chapter "Hardware Administration").
- ***Administrator Capability*** - If this field is set to "Y es", the user is able to send broadcast messages, record a custom call answering greeting and personal verifications for all other users. The field defaults to the value set in the user model.

- ***Logon Status*** - When the status is “Disabled”, the user cannot log on to the system, however, messages are still received. This may happen if too many logon attempts are made using the wrong password. If the status is “Disabled” an explanation is displayed on the line below this field. When the status is “Enabled” the user has full access to the mailbox. The default is “Enabled”.
- ***Billing Class*** - The user’s billing classification determines the rate at which the user is billed for DMS VoiceMail voice services. This information appears in the User Usage Report and is used by host-based billing applications through file download via AdminPlus. (See the *AdminPlus System Administration Guide*.) The valid range is from 0 to 7. The field defaults to the value set in the user model.
- ***Read Message Retention*** - This field allows you to specify the number of days that messages are kept in the user’s mailbox after they have been read. Once the lesser of this number or the organization’s maximum read message retention limit (see the “Voice Messaging Options” section in the “Voice Administration” chapter for details) is reached, read messages are automatically deleted. If “0” is entered for both values, read messages are not automatically deleted by the system, but can only be deleted by the user. The field defaults to the value set in the user model.

  The value you enter here is limited by the system-wide value set in the *Max Read Message Retention* field in the Voice Messaging Options screen. (See “Voice Messaging Options” in the chapter “Voice Administration”.) The following table explains which value is used to determine how long the user’s read messages are kept.

System Retention Limit	User Retention Limit	Amount of Time Read Messages are Kept
0 (zero)	0 (zero)	Messages are kept until the user deletes them. The system will not automatically delete read messages.
0 (zero)	A non-zero value	The user retention limit determines how long messages are kept.
A non-zero value	0 (zero)	The system retention limit determines how long messages are kept.
A non-zero value	A non-zero value	The lesser value is used to determine how long messages are kept.

- ***Retain Copy of Sent Messages*** - When this field is set to “Yes”, copies of sent messages are not deleted from the user’s mailbox. When it is set to “No”, messages are deleted as soon as they are sent. To minimize use of storage, use “No”. The field defaults to the value set in the user model.

- ***Auto Logon*** - When this field is set to “Y es”, the user does not need to enter a mailbox number or password to gain access to DMS VoiceMail. When set to “No” the user must enter a mailbox number and password. Use “Yes” only for users with telephones in secure locations. The default is the same as the value set in the user model.
- ***Delayed Prompts*** - When this field is set to “Y es”, the system will prompt the user for an action if the user does not initiate any action for 3.5 seconds. Set this field to “Yes” for a new user. If this is an experienced user, you may want to set this field to “No”. The default is taken from the user model.
- ***Auto Play*** - When set to “Yes”, the messages in the user’s mailbox are automatically played when the user logs on, starting from the first new message, and then from the old messages when the last new message is played. When set to “No”, the user must explicitly request that each message be played by pressing “2” on the telephone keypad. The field defaults to the value set in the user model.
- ***Callers Notified of Busy Line*** - Set this field to “Yes” if this user’s callers are to be informed if the called line is busy. After being so informed, the caller is connected to DMS VoiceMail. If the field is set to “No”, the caller is simply connected to DMS VoiceMail and given the chance to leave a message.

  Note: If the user’s mailbox is associated with two (or three) DNs, they must all be busy for this prompt to be played.

- ***Dual Language Prompting*** - This field is displayed on multilingual systems only. The selection made here affects the prompts played to people calling from external phones. (It does not apply to the prompts played to DMS VoiceMail users. The language in which prompts are played to users who are logged on to DMS VoiceMail is determined by the following field, *Preferred Language*.)

  If this field is “Enabled”, callers hear prompts in the user’s preferred language (as specified in the next field), followed by the primary default language. If the primary default language is the same as the preferred language, the secondary default language (as specified in the Voice Messaging Options screen) will follow the preferred language. If this field is set to “Disabled”, prompts are played only in the primary default language.

  When this field is set to “Enabled”, prompts are played in the primary default (system) language, followed by the secondary default language (as specified in the Voice Messaging Options screen). If this field is set to “Disabled”, prompts are played only in the primary default language.

- ***Preferred Language*** - This field applies only to multilingual systems. The language specified in this field determines the language in which prompts are played (this includes prompts that are played to the user during a login session and to callers during express messaging and call answering sessions). This field can display a maximum of four of the languages installed on your system. If you do not specify a language, the language that was first installed is used.

  Note: If *Default Language Overrides User's Preferred Language* is set to "Yes" in the Voice Messaging Options screen, prompts played during call answering and express messaging sessions will be in the default language.

- ***Custom Revert Restriction/Permission Codes*** - The custom revert DN is the extension to which a caller is passed when the caller presses 0 during a call answering session. Since users can customize this DN from their telephone set you must determine which dialing codes you want to restrict (or explicitly permit). For example, you may want to ensure that users cannot revert callers to certain extensions within the company (such as the President's office, etc.)

  The actual restriction/permission tables are defined in the Voice Security Options screen (described in the "Voice Administration" chapter). Up to 10 restriction and 10 permission codes can be defined for each option. The default option is "Local".

  Note: If "None" is selected, all dialing codes are permitted since no specific restrictions are applied.

- ***Extension Dialing Restriction/Permission Codes*** - This field indicates which restricted/permitted dialing codes apply when the user dials a phone number while logged on to his mailbox (known as mailbox thru-dialing). For example, a user may dial into the office from an external trunk in order to listen to messages. While listening to messages he realizes he would like to speak to someone at the office. Instead of logging out and calling back, the user can press "0" followed by the extension number. You may want to restrict users from dialing external or long distance numbers when thru-dialing. The four choices displayed in this screen reflect the four sets of dialing codes that have been defined in the Voice Security Options screen (described in the chapter "Voice Administration"). Each set contains up to 10 permission and 10 restriction codes.

  Note: If "None" is selected, all dialing codes are permitted since no specific restrictions are applied.

- ***External Call-Sender Restriction/Permission Codes* -** This field is only applicable if Meridian Networking is enabled. (Meridian Networking is only available on CPE systems.) When a message is left in a user's mailbox during a call answering session, the External Call Sender feature allows the user to call the sender immediately after listening to the message (by pressing "9"). Note that this only applies to users sending messages from remote DMS VoiceMail Network sites who are not defined as remote voice users in the local system. If the sender of a message is defined as a remote voice user, then these restriction/permission codes are not checked and all calls are permitted. (Note that this feature only applies to messages left during call answering sessions, not voice messages that have been composed and sent. The Reply To feature applies to voice messages.)

  You may want to restrict users from using this feature to dial certain long distance DNs (or other specified external DNs). To do so, select the appropriate restriction/permission codes. The four choices that are presented in this screen reflect the four sets of dialing codes that are defined in the Voice Security Options screen. If "None" is selected, all dialing codes are permitted since no specific restrictions are applied.
- ***Receive AMIS Messages* -** This field is displayed only if AMIS is installed. If the field is set to "Yes" users can receive AMIS messages.
- ***Compose/send AMIS Messages* -** This field is displayed only if AMIS is installed. If the field is set to "Yes", users can compose and send AMIS messages.
- ***AMIS Restriction/Permission Codes* -** This field is displayed only if AMIS is installed and the previous field, *Compose/send AMIS messages*, is set to "Yes". The field indicates which restricted/permitted dialing codes apply when a user sends an AMIS message. If "None" is selected, all dialing codes are permitted since no specific restrictions are applied.

Figure 6-6xxx
Add Local Voice User (Change Defaults) for the VMUIF interface

```
User Administration

Add Local Voice User
Mailbox Number:     8765432          Volume ID:  203
Storage Limit (minutes): 123         Storage Used:  12
Last Name:     ___________________________________
First Name:    _______________________  Initials: ________
Extension DNs: 8765432_____________
               ____________________
               ____________________
Revert DN:     ____________________
Message Waiting Indication Option: None [Any]  Urgent
*Message Waiting Indication DN:   ____________________________
*Message Waiting Link name:       ____________________________
Personal Verification Recorded (Voice): No

                                                     MORE BELOW
```

* These fields are displayed only if MWI Option is set to “Any” or “Urgent”.

Figure continued on next page

Figure 6-6 (continued)
Add Local Voice User (Change Defaults) for the VMUIF interface

```
Add Local Voice User                                         MORE ABOVE

Logon Status:                          Disabled  [Enabled]

Lockout Duration (hh:mm):              00:00
(00:00 implies no mailbox reset)

Billing Class:                         2

Read Message Retention (days):         10
("0" implies that read messages
are retained until the user
deletes them manually.)
(Subject to the system's
maximum retention of 7 days)

Dial Pulse Support:                  [No] Yes

Auto Logon:                          [No] Yes

Delayed Prompts:                     No  [Yes]

                                                             MORE BELOW
```

```
Add Local Voice User                                         MORE ABOVE

   Skip to First New Message:          [No] Yes

   Callers Notified of Busy Line:      No  [Yes]

 * Preferred Language:                 [AmericanEnglish]
                                       EuropeanEnglish
                                       Mandarin

   Custom Revert                       None  On_Switch [Local] Long_distance_1
   restriction/permission codes:       Long_distance_2

** Receive AMIS messages:              [No]   Yes

Save    Cancel    Outcalling Fields    Voice    Change Password
```

* This field is only displayed on multi-lingual systems.
**This field is only displayed if AMIS networking is installed.

When you press the [Change Defaults] softkey, the fields in the first Add a Local Voice User screen are displayed again. They are filled in with the values that you entered before pressing the [Change Defaults] softkey. These fields have already been described in the previous section "Adding local voice users" and will not be described here.

The following fields are displayed if the Call Answering feature and the VMUIF interface are enabled. Some of these fields are prefilled with default settings. These defaults are based on the user model upon which the user profile is based. User Models are discussed on page 6-87.

- ***Storage Limit (minutes)*** - The maximum amount of storage available to the subscriber. You may enter a value from 1 to 360 (minutes). This field defaults to the value set in the user model.

  If a subscriber surpasses this limit, calls are not cut off. Instead, when the subscriber logs on, he or she hears a message indicating that the mailbox is full and is prompted to delete messages.

- ***Storage Used*** - This read-only field indicates how many minutes of voice messages are currently stored for the current subscriber. This value is rounded up to the nearest minute. Before deleting a subscriber, check this field to make sure that there are no voice messages in the mailbox.

  Note: A user may inform you that he or she has received the mailbox full warning, but that the mailbox is definitely not full. For example, the user is certain that there are only two short messages in the mailbox. A prematurely full mailbox is caused by an unexpected system reboot that leaves inconsistencies between the volume server and what is actually in the mailbox. This problem will be fixed automatically during the scheduled nightly audit. However, if an unexpected reboot happens at a busy traffic time, you can log on at the Tools level and select the menu item "Audit all volumes". This will update the real mailbox storage information that is stored on disk and prevent prematurely full mailboxes. See the *System Administration Tools Guide* (NTP 555-7001-305) for more information about this tool.

- ***Message Waiting Indication Option*** - The chosen setting determines whether or not subscribers are notified of new messages. If you select "Any" or "Urgent" the feature will be turned on and subscribers will be notified of new messages that have been received in their mailbox. The notification is in the form of an interrupted dial tone or a flashing light (if the subscriber has a phone with a message waiting light.) If you select "None", this essentially turns message waiting indication off. (This option should be selected if the user has a mailbox but no physical telephone set.) The field defaults to the value set in the user model. The following two fields (*Message Waiting Indication DN* and *Message Waiting SMDI Link Name*) are not displayed if "None" is selected.

- ***Message Waiting Indication DN*** - This field is displayed only if the *Message Waiting Indication Option* field is set to “Any” or “Urgent”. This DN specifies the telephone extension at which message waiting indication is activated when a new message is put in the user’s mailbox. This field defaults to the user’s mailbox number. Set this DN to the user’s 7-digit directory number as it is configured on the switch.
- ***Message Waiting Link Name -*** This field is displayed only if *Message Waiting Indication Option* is set to “Any” or “Urgent”. It specifies the link on which the message waiting indication is sent for this user.

  This field is intended for systems with the Multi-SMDI (referred to as “SMDI” in the General Options screen) feature so that you can distribute users over all available links. If you have only one SMDI link, this field defaults to the link name entered in the hardware database and cannot be changed from this screen.

  If you do have multiple SMDI links, do not put all users on the same link. Instead, distribute users (as evenly as possible) across all available links. This field defaults to the first link name defined in the hardware database. You can cycle through the other link names while in this screen. To display the next link name, delete the current link name and press <Tab> or <Return>. Repeat this procedure until you have cycled through all of the link names (you may want to write them down as you go). You can also view the the link names in the View Data Port screen (SMDI), as described in the chapter “Hardware Administration”).
- ***Logon Status*** - When the status is “Disabled”, the subscriber cannot log on to the system, however, messages are still received. This may happen if too many logon attempts are made using the wrong password. If the status is “Disabled” an explanation is displayed on the line below this field. When the status is “Enabled” the subscriber has full access to the mailbox. The default is “Enabled”.
- ***Lockout Duration (hh:mm) -*** When a subscriber’s mailbox is disabled due to password violation, this field determines how long the subscriber is locked out before he can log on again. You may enter a value from 00:00 to 23:59. If you enter a value of 00:00, this means that the subscriber will be locked out until you (the administrator) decide to re-enable the mailbox.
- ***Billing Class*** - The subscriber’s billing classification determines the rate at which the subscriber is billed for DMS VoiceMail voice services. This information appears in the User Usage Report and is used by host-based billing applications through file download via AdminPlus. (See the *AdminPlus System Administration Guide*.) The valid range is from 0 to 7. The field defaults to the value set in the user model.

- ***Read Message Retention*** - This field allows you to specify the number of days that messages are kept in the subscriber's mailbox after they have been read. Once the lesser of this number or the organization's maximum read message retention limit (see the "Voice Messaging Options" section in the "Voice Administration" chapter for details) is reached, read messages are automatically deleted. If "0" is entered for both values, read messages are not automatically deleted by the system, but can only be deleted by the subscriber. The field defaults to the value set in the user model.

  The value you enter here is limited by the system-wide value set in the *Max Read Message Retention* field in the Voice Messaging Options screen. (See "Voice Messaging Options" in the chapter "Voice Administration".) The following table explains which value is used to determine how long the subscriber's read messages are kept.

System Retention Limit	User Retention Limit	Amount of Time Read Messages are Kept
0 (zero)	0 (zero)	Messages are kept until the user deletes them. The system will not automatically delete read messages.
0 (zero)	A non-zero value	The user retention limit determines how long messages are kept.
A non-zero value	0 (zero)	The system retention limit determines how long messages are kept.
A non-zero value	A non-zero value	The lesser value is used to determine how long messages are kept.

  Note: It is important to configure the proper message retention value for call answering subscribers that have dial pulse phones since automatic read message deletion is the only way for them to remove messages from mailboxes.

- ***Dial Pulse Support -*** Set this field to "Yes" if the subscriber has a rotary or dial pulse phone. Set this field to "No" if the subscriber has a touch-tone phone. If this field is set to "Yes", Auto Logon (the next field) must also be set to "Yes".
- ***Auto Logon*** - When this field is set to "Yes", the subscriber does not need to enter a mailbox number or password to gain access to DMS VoiceMail. When set to "No" the subscriber must enter a mailbox number and password.

You will have to enable auto logon for subscribers with rotary phones since they will not be able to enter a password through a keypad. You can also set this field to “Yes” if the subscriber prefers a simpler interface and does not want to have to enter a password. However, this feature can not be enabled if the subscriber wants to be able to log on to his or her mailbox from a remote phone or if he or she wants remote notification capabilities. The default is the same as the value set in the user model.

- ***Delayed Prompts*** - When this field is set to “Yes”, the system will prompt the subscriber for an action if there has been no activity for 3.5 seconds. Set this field to “Yes” for a new subscriber. Experienced subscribers may find delayed prompts unnecessary and may prefer to have this feature disabled. The default is taken from the user model.
- ***Skip to First New Message -*** This field determines what happens when a subscriber logs on to listen to new messages. If this field is set to “Yes”, the first new message is automatically played when the subscriber successfully logs on. If this field is set to “No”, subscribers must use the Play command to listen to new messages.
- ***Callers Notified of Busy Line -*** Set this field to “Yes” if this subscriber’s callers are to be informed if the called line is busy. After being so informed, the caller is connected to DMS VoiceMail. If the field is set to “No”, the caller is simply connected to DMS VoiceMail and given the chance to leave a message.
- ***Preferred Language*** - This field applies only to multilingual systems . The language specified in this field determines the language in which prompts are played (this includes prompts that are played to subscribers while they are logged into their mailbox as well as to callers during call answering sessions). This field can display a maximum of four of the languages installed on your system. If you do not specify a language, the language that was first installed is used.

  Note: If *Default Language Overrides User’s Preferred Language* is set to “Yes” in the Voice Messaging Options screen, prompts that are played during call answering and express messaging sessions are in the default language.

- ***Custom Revert Restriction/Permission Codes*** - The custom revert determines the DN to which a caller is passed when the caller presses 0 during a Call Answering session.

  The actual restriction/permission tables are defined in the Voice Security Options screen (described in the “Voice Administration” chapter). Up to 10 restriction and 10 permission codes can be defined for each option. The default option is “Local”.

- ***Receive AMIS Messages -*** This field is displayed only if AMIS is installed. If the field is set to “Yes” users can receive AMIS messages.

Note: Call Answering subscribers can receive AMIS messages but can not compose or send them. It is therefore unnecessary to configure restriction and permission codes for this service.

Procedure 6-6xxx
Changing defaults for local voice users

Starting point : User Administration screen, [Add Local Voice User] entered.

1 Press [Change Defaults].

The additional default fields are accessible (Figure 6-5). Use the cursor to move to the bottom of the screen and scroll upwards.

2 Move the cursor to the field to be changed.

3 Enter the data, or select an option, as required.

4 Use the [Outcalling Fields] softkey to access Remote Notification and Delivery to Non-Users parameters.

5 If a personal verification has not been recorded, use [Voice] to record one.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Record] softkey.

c. At the sound of the beep, speak the user’s name into the telephone handset.

d. Press the [Stop] softkey to stop recording.

e. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the recording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

See the section, “Recording personal verifications using the [Voice] softkey” on page 6-56 for more information about the recording softkeys.

6 Use [Change Password] to set the user’s DMS VoiceMail password.

You are prompted to enter the new password, then to re-enter the new password to verify it. The passwords are not displayed on the screen as you type them.

User passwords must be numeric and can be up to 16 digits long. By default, the initial password for a new user is the same as the user’s mailbox number. For Call Answering subscribers, there is no initial password.

7 Go to step 7a to save the new user, or 7b to cancel the addition.

a. Use [Save].

The system saves the new user, including any changes to the Change Defaults and Outcalling fields, and prompts for another local voice user's mailbox number. Go to "Adding a Local Voice User" earlier in this chapter, to add another user, or to 7b to leave the menu.

b. Use [Cancel].

All new user information, including any changes to the Change Defaults or Outcalling fields, is discarded. The Add New User softkeys are re-displayed; see "Adding new users" earlier in this chapter for details.

Defining Outcalling parameters

The Outcalling feature includes Remote Notification (RN) and Delivery to Non-Users (DNU). Remote Notification (RN) informs users via a remote telephone, pager, or paging service, that there are new messages in their mailbox. Delivery to Non-Users allows an MMUI DMS VoiceMail user to compose and send messages to non-users of DMS VoiceMail. A number of RN and DNU parameters are set on a per-user basis in one of the following User Administration screens: Add Local Voice User, View/Modify Local Voice User or View/Modify User Models (described later in this chapter). Other RN and DNU parameters are set on a system-wide basis in the Outcalling Options screen which is accessible from the Voice Administration menu (described in the “Voice Administration” chapter).

> ***Note:*** Because call answering subscribers cannot compose and send voice messages, DNU does not apply when the VMUIF interface is enabled.

Remote Notification

Voice messaging (MMUI) users can be notified of any new messages that arrive in their mailbox or only those messages that are tagged as urgent. This is determined in the *Message Remote Notification Options* field in the Outcalling Fields in the Add or View/Modify a Local Voice User screen. This can also be specified by the user if he or she is creating his or her own remote notification schedule. Call answering (VMUIF) subscribers are notified of all new messages since call answering messages cannot be tagged as urgent.

When the user answers the target number, the remote notification service plays a message indicating that messages have been received. Users are then asked to log in and listen to messages (by pressing “1” on their telephone keypad), or to turn off remote notification (by pressing “3”).

> ***Note for Call Answering subscribers:*** Ensure that user passwords have been created for those subscribers that need remote notification capability.

RN Target DNs

Remote Notification allows each user to have three separate time periods in a 24-hour day, each with up to three target numbers. These target numbers can be pager service numbers, local or long distance numbers. They can be up to 30 digits in length. You can also specify which numbers are restricted or permitted for each user.

WARNING
Do not enter the user’s own extension as the RN target DN
This will cause a cyclical build-up of messages in the user’s mailbox until the disk is full. The retry repeat cycle is not halted because each retry repeat causes a new message to be sent, which in turn starts remote notification all over again for the new messages.

Schedules

The administrator, using the administration terminal, can set up Remote Notification schedules for each user. The administrator can also give Voice Messaging users the capability to set up their own schedules using their telephone keypads by enabling the *Keypad Interface* field in the Outcalling Fields in the Add and View/Modify Local Voice User screens. (Call Answering subscribers cannot create schedules from their telephone sets.) Figure 6-8 on page 6-49 displays the fields used to create schedules.

A schedule allows you to define numbers where users can be reached at different times of the business day, as well as non-business days. There are three different schedules associated with each user: one for business days, one for non-business days, and one temporary schedule. The temporary schedule overrides the other two schedules until the time specified. This schedule is useful if a user will be at a different number for a short period.

Each schedule is subdivided into three time periods. This allows you to associated different numbers with different times of the day. For example, a user may typically be at a particular client’s office during the morning, but on the road during the afternoon. You can therefore, direct Remote Notification to call the client’s office between 9:00 a.m. and 12:00 p.m., and call the user’s car phone between 12:01 and 5:00 p.m. (Beginning one time period one minute after the previous time period carries RNs over from the first time period. If there is a gap between time periods, RNs initiated in one time period are dropped once the time period ends.)

Each time period within a schedule can have up to three target DNs associated with it. When there is more than one target number, DMS VoiceMail calls the first DN. If that phone or pager was unanswered or answered without the recipient logging in, the next DN is tried. If the user answers the phone and logs in, RN stops regardless of whether or not the user has listened to the new message. However if while the user is logged in, a new message comes in that is not announced, RNs will still continue to be sent. Normally, new messages are announced, but they will not be announced if the user disconnects while listening to an existing message.

When RN has stopped for a particular message (or messages), a new message will reactivate RN.

Retry Limits and Intervals

When a Remote Notification attempt is unsuccessful, DMS VoiceMail uses Retry Limits to determine how many times it should attempt to contact the user. Retry Intervals determine how often the retries should be attempted (i.e., the amount of time between retries, from 00:00 to 23:59 (mm:ss)).

There are three types of unsuccessful RN attempts: Busy, No Answer, and Answer No Login (where the user answers the phone, or pager, but does not log in). Each of these three conditions has a Retry Limit and Retry Interval associated with it. See Figure 6-7 on page 6-46.

Remote Notification Retry Scenarios

The RN retry sequences that result depend on the type of unsuccessful RN attempt (Busy, No Answer, Answer No Login), and whether there is only one target DN or multiple targets specified in the time period. The first scenario looks at a situation in which only one target DN is specified for the time period, whereas in Scenario 2 there are three target DNs associated with the time period.

In general, the following rules apply when there is one target DN. If a call is:

- ***Answered***, but there is no login, (and no explicit request to turn RN off), the Answered Retry Limit is used. (If the user logs in but does not listen to any messages, remote notification is turned off.)

  Although users are likely to either log in or turn remote notification off, there are several situations in which this “answer no login” scenario can arise. For example, the remote notification service calls a phone that is connected to an answering machine. The RN service will play the message prompting the user to log in or turn RN off, yet there will be no action (no login, RN is not turned off). Remote notification will continue to call the target DN the number of times specified by the Answered Retry Limit.

  An “answer no login” may also occur if the target DN is a pager. DMS VoiceMail will make a call to the pager, which will answer the call and may receive data (such as the callback number). The call will then be disconnected. If the user does not respond to the pager by logging on to his or her mailbox, RN will continue.
- ***Not Answered***, the No Answer Retry Limit and Retry Interval are followed.
- ***Busy***, then the Busy Retry Limit and Retry Interval are followed until exhausted. Once the Busy Retry Limit is exhausted, RN will continue to call using the No Answer Retry Limit and Retry Interval. Once exhausted, Remote Notifications stops.

Scenario 1: 1 target DN defined for the first time period (9:00 a.m. to 12:00 p.m.) in a business day schedule

Busy Retry Limit = 3 and Interval = 5 mins
No Answer Retry Limit = 10 and Interval = 15 mins
Answered No Login Retry Limit = 1 and Interval = 5 mins

Table 6-1xxx
RN retry scenario for a schedule time period with one target DN

Time of Message	RN Action	RN Result	Further Action
8:55 a.m. message arrives	No RN sent - this is before the first time period	None	None
9:30 a.m. message arrives	RN sent	Busy	RN rescheduled using Busy Retry Limit and Interval
9:35 a.m.	First Busy Retry	Busy	RN rescheduled using Busy Retry Limit and Interval
9:40 a.m.	Second Busy Retry	No Answer	RN rescheduled using No Answer Retry Limit and Interval
9:55 a.m.	First No Answer Retry	Busy	RN rescheduled using Busy Retry Limit and Interval
10:00 a.m.	Third Busy Retry	Busy	Busy Retry Limit is exhausted; RN rescheduled No Answer Retry Limit and Interval
10:15 a.m.	Second No Answer Retry	No Answer	RN rescheduled using No Answer Retry Limit
10:20 a.m. message arrives	*		
10:30 a.m.	Third No Answer Retry	Answered No Login	RN rescheduled using Answered No Login Retry Limit and Interval
10:35 a.m.	First Answered No Login Retry	Answered No Login	Answered No Login Retry Limit exhausted; RN stops until a new message arrives
11:52 message arrives	RN sent	Busy	RN rescheduled using Busy Retry Limit and Interval
11:57 a.m.	First Busy Retry	Busy	The RN retry falls outside of the time period. RN stops.

*While within an RN cycle (a series of retries initiated by a new message arriving in a mailbox), new messages do not initiate a new notification attempt. The first retry cycle is used to notify the user of all messages. A message initiates RN only when there is no RN cycle currently in progress.

When the retry limits have been exhausted, Remote Notification stops until another new message is deposited into the user's mailbox. Further limits are placed on the number of *retry cycles* - a cycle refers to one pass through the number of allowed retries. See the description of the *Maximum Number of Remote Notification Retry Repeats* field in the section “Outcalling Options” the “Voice Administration” chapter.

When there is more than one target DN defined for a time period these additional rules apply. When a retry cycle is first initiated, DMS VoiceMail calls the first DN. If that call is:

- ***Answered*** (with no login) or ***Not Answered***, the next DN is called.
- ***Busy***, the next DN is not called. Instead, the same DN is called using the Busy Retry Limit and Interval. If on a subsequent retry, the busy number becomes No Answer or Answered without Login, the next target DN is called.
- When the response to a call at one DN is different from that at another DN (for example, No Answer at target DN 1, Answer without Login at target DN 2, and No Answer at target DN 3) the “best” call result is used to determine which retry limit and interval should be used. Results are preferred in the following order:

  1. Answered without Login (a pager or an answering machine has answered the phone)
  2. No Answer (the person is probably not at the phone)

  (Remember that when a Busy condition is encountered, DMS VoiceMail does not go to the next DN. If a Busy condition is present, it is always preferred.)

Scenario 2: multiple (3) target DNs defined for the first time period (9:00 a.m. to 12:00 p.m.) in a business day schedule
Busy Retry Limit = 3 and Interval = 5 mins
No Answer Retry Limit = 10 and Interval = 15 mins
Answered No Login Retry Limit = 1 and Interval = 5 mins

> ***Note:*** One cycle through a set of three target DNs is considered one call in terms of retries.

Table 6-2xxx
RN retry scenarios for a schedule time period with multiple target DNs

Time of Message	RN Action	RN Result	Further Action
9:10 a.m. message arrives	RN sent to Target DN 1	No Answer	Next target DN is called
	RN sent to Target DN 2	Answered No Login	Next target DN is called
	RN sent to Target DN 3	Answered with Login	RN stops
9:20 a.m. message arrives	RN sent to Target DN 1	No Answer	Next target DN is called
	RN sent to Target DN 2	No Answer	Next target DN is called
	RN sent to Target DN 3	Answered No Login	The Answered No Login Retry Limit and Interval are used (this was the best call)
9:25 a.m.	First Answered No Login retry sent to Target DN 1	Busy	RN is rescheduled using the Busy Retry Limit and Interval. The same target DN will be called.
9:30 a.m.	First Busy retry sent to Target DN 1	No Answer	Next target DN is called
	RN sent to Target DN 2	Busy	RN is rescheduled using the Busy Retry Limit and Interval.
9:35 a.m.	Second Busy retry sent to Target DN 2	Busy	RN is rescheduled using the Busy Retry Limit and Interval
9:40 a.m.	Third Busy retry sent to Target DN 2	Busy	Busy Retry Limit is exhausted; reschedule call to same DN using No Answer Retry Limit
9:55 a.m.	First No Answer retry sent to Target DN 2	No Answer	Next target DN called
	RN sent to Target DN 3	No Answer	RN rescheduled using No Answer Retry Limit and Interval
10:10 a.m.	Second No Answer retry sent to Target DN 1	Answered with Login	RN stops

Sending remote notifications to pagers

The remote notification feature can make calls to the following types of pagers:

- Tone-only pager
- Tone and Voice pager
- Digital or Numeric (display) pager
- General access pager service

Pager services typically provide one of two pager activation methods to subscribers. Services which cater to a local market tend to offer DID numbers for pager activation. In this case, each pager is assigned a unique DID number. Tone-only and Tone and Voice Pagers are almost always offered with DID numbers. Services which cater to large regional or national markets tend to offer 800 numbers.

With general access pager services, all pagers share a common local or 800 number. DMS VoiceMail dials this number followed by the PIN number of the pager. DMS VoiceMail then sends a call-back number which is displayed on digital display pagers. This is the number the user calls to retrieve his or her messages (usually the DMS VoiceMail voice messaging DN). The entry of the PIN number and call-back numbers are usually terminated by the # character. However, some services use fixed length PIN numbers and do not accept a terminator character.

Certain requirements must be met for DMS VoiceMail to work properly with the supported pagers. For a remote notification message to be delivered successfully, DMS VoiceMail must recognize that the paging company has responded to its call. A call is considered answered under the following conditions.

1. There is voice detection. This is the preferred method.

   If voice is detected, DMS VoiceMail will wait a maximum of 20 seconds for silence to be detected. When silence is detected, or when the timeout period expires, DMS VoiceMail will continue with notification delivery.

2. There is tone detection. This tone can have one of the following frequencies:

   a. 1400 Hz - the Northern American standard frequency. This tone must have a maximum duration of 3.5 seconds.

   b. 1000 Hz - only if a minimum of two on-off cycles are presented and the maximum duration is 5.0 seconds. Call your Northern Telecom support organization if it is necessary to extend this pager tone recognition time.

3 The paging company answers the call only AFTER the calling side (DMS VoiceMail) has been allowed to hear the ringback tone cycle two times. At this point, a tone or voice prompt may be provided. Using this method, the frequency of the answering tone is no longer important, but the timing of the DMS VoiceMail interaction will be delayed. The service must be prepared to wait for seven seconds after it has responded with an answering signal before receiving a reply from DMS VoiceMail.

If a pager fails to respond to a remote notification call, call the paging company to ensure that it meets one of the above requirements.

Once there is a tone or voice prompt in response to the remote notification call, the service must be immediately ready to accept the DMS VoiceMail response. What is received from DMS VoiceMail depends on the pager type that has been specified in the Add a Local Voice User screen. See the following sections for more details.

Tone-only pagers and tone and voice pagers

Both tone-only and tone and voice pagers are handled in the same way by DMS VoiceMail. DMS VoiceMail plays the notification prompt immediately after recognizing that the call has been answered.

Numeric pagers

With a display or numeric pager, DMS VoiceMail sends a callback number and a pager data terminator. The callback number is the number the user must dial to retrieve new messages, and is usually the DMS VoiceMail access number. The callback number may consist of up to 8 characters, using the decimal digits “0-9” and the asterisk. DMS VoiceMail gets the callback number from one of two places: either the user profile (from the remote notification schedule) or, if it is not defined for the user, from the Outcalling Options screen (see the “Voice Administration” chapter).

Some paging companies require a pager data terminator character. This terminator character is defined in the Outcalling Options screen and is used to terminate the callback number. The pager data terminator will either be the pound sign (#) or nil (if not required). There is no support for a pager prefix.

After DMS VoiceMail recognizes that a notification call has been answered, it waits two seconds. If a callback number and/or pager data terminator are defined, they are outpulsed and there is a three second delay. A voice prompt is then played to notify the paging company that a message has been received in the user’s mailbox.

General access pager services

When remote notification is sent to a general access pager service, you must define the pager ID in the Add or View/Modify Local Voice User screen (Outcalling Fields) where you normally define the callback number. You, therefore, cannot customize the callback number for each user. Instead, DMS VoiceMail gets the callback number from the Outcalling Options screen. (See the “Voice Administration” chapter.)

The pager data terminator is defined in the Outcalling Options screen and is used to terminate both the pager ID and the callback number. This terminator can either be the pound sign (#) or nil (since not all pager services use a terminator character). There is no support for a pager prefix.

After DMS VoiceMail recognizes that a notification call has been received, it waits two seconds before the pager ID and the pager data terminator are outpulsed. DMS VoiceMail then waits for the paging company to answer with voice or tone. When DMS VoiceMail receives an answer, there is a two second delay. If a callback number and/or a pager data terminator are defined, they are outpulsed and there is a three second delay. A voice prompt is then played to notify the paging company that a message has been received in the user’s mailbox.

Delivery to Non-Users

This feature enables Voice Messaging users to compose and send messages to people who do not subscribe to a DMS VoiceMail system. It is not available to Call Answering subscribers since they do not have the ability to compose and send messages. Like Remote Notification, DMS VoiceMail uses retry scenarios when attempting to deliver messages to non-users.

Message playback to a non-user can be trigger by one of two things: *DTMF confirmation* or voice detection. (Your preference is defined in the Outcalling Options screen, described later in this chapter.) If DTMF confirmation is enabled, DMS VoiceMail plays a message to the recipient, prompting him or her to press 2 if they want to hear the message from the DMS VoiceMail user, or to hang up if they do not want to take the call. This safeguards against the wrong person receiving the message. However, if there are many rotary phones in your area, it is not recommended that you enable DTMF confirmation since recipients will not be able to press 2. If you specify that DTMF confirmation is not required, the message is automatically delivered when the call is answered and voice is detected.

After a recipient has listened to a message, the non-user can record a reply back to the sender. If the non-user does not record a reply, and the original message was tagged for acknowledgement, a reply, in the form of a system acknowledgment, will be sent to the originator of the message.

Most DNU parameters are specified on a system-wide basis in the Outcalling Options screen described in the “Voice Administration” chapter. These include:

- The time during weekdays and weekends that messages are allowed to be delivered to non-users (it is important to know the restrictions on electronic delivery of phone messages that apply to your geographical region).
- The number of times that the DNU service re-dials the non-user number when the called number is busy, unanswered, or 2 is not pressed (when DTMF confirmation is required).
- Addressing prefixes and associated dialing codes - the prefix indicates to DMS VoiceMail that the number about to be dialed is the number of a non-user and not a mailbox and the dialing code is the number actually dialed (e.g., 9 to access an outside line).
- The number of times to play a message to a non-user
- Whether or not DTMF confirmation is required.
- Whether or not users’ preferences for DTMF confirmation should be overrided by the system setting.

User-specific parameters are described in the following sections. These include:

- Whether or not the user has DNU capability. The default is “No”.
- Whether or not DTMF confirmation is required.
- The dialing codes that are restricted and/or permitted for the user.

Filling in Outcalling Fields for Local Voice Users

If the Outcalling feature is configured on your system, the [Outcalling Fields] softkey appears on the Add Local Voice User - Change Defaults screen. When you press the [Outcalling Fields] softkey, fields associated with the Delivery to Non-Users and Remote Notification features (Figure 6-7) are displayed. Several other Outcalling parameters are set on a system-wide basis. These are described in “Outcalling Administration” in the “Voice Administration” chapter.

Figure 6-7xxx
Add Local Voice User (Outcalling Fields)

```
                                User Administration

  Add Local Voice User - Outcalling Fields

 *Delivery to Non-Users Capability:          No   [Yes]

 *DNU DTMF confirmation required:            [Disabled] Enabled

 *Delivery to non-user                       None  On_Switch [Local] Long_distance_1
  restriction/permission codes               Long_distance_2

                     -----------------------

  Remote Notification Capability:            [No] Yes

  Current State:                             Off

 *Keypad Interface:                          Disabled  [Enabled]

 *Message Remote Notification Option:        [Any]  Urgent

  Remote Notification                        None  On_Switch [Local] Long_distance_1
  restriction/permission codes:              Long_distance_2

  Busy      Retry Limit:3__     Retry Interval (hh:mm):  00:05
  No Answer Retry Limit:10_     Retry Interval (hh:mm):  00:15
  Answered  Retry Limit:1__     Retry Interval (hh:mm):  00:05

  Business Days:      Sunday                 [No]  Yes
                      Monday                  No  [Yes]
                      Tuesday                 No  [Yes]
                      Wednesday               No  [Yes]
                      Thursday                No  [Yes]
                      Friday                  No  [Yes]
                      Saturday               [No]  Yes

                                                                       MORE BELOW
```

* These fields are not displayed if Call Answering (VMUIF) is installed on the system.

Figure continued on page 6-49.

The following fields appear on the Outcalling Fields screen:

- ***Delivery to Non-Users Capability*** - This field is applicable only if Voice Messaging (MMUI) is enabled. This field determines whether or not this subscriber can compose and send messages to people who are not subscribers of a DMS VoiceMail system. The default is “No”.
- ***DNU DTMF Confirmation Required*** - This field is applicable only if Voice Messaging (MMUI) is enabled. This field indicates whether or not a recipient of a Delivery to Non-user (DNU) message is required to confirm that they want to hear the message by pressing **2**. This can help avoid messages being delivered to an answering machine or to the wrong person. When disabled, the message is played upon voice detection. If you are in an area where rotary phones are widely used, you should disable confirmation. The default is “Disabled”.
- ***Delivery to Non-User Restriction/Permission Codes*** - This field is applicable only if Voice Messaging (MMUI) is enabled. If DNU capability is disabled for the user, this field is not displayed. The selected option determines which dialing codes can and cannot be dialed when this user attempts to send a message to a non-user. The actual dialing codes are defined in the Voice Security Options screen, accessible through the Voice Administration menu.
- ***Remote Notification Capability*** - This field determines whether or not this user can be notified at a remote telephone or pager of messages waiting in his mailbox. The default is “No”. If this field is set to “No” the rest of the fields on this screen are not relevant.
- ***Current State*** - This read-only field indicates whether the Remote Notification service is currently active for this user. The options are: On/Off, Off by Retry, Off by Called Party and Off due to Bad DN.
  - ***On/Off*** indicates that the administrator has enabled or disabled Remote Notification.
  - ***Off by Retry*** indicates that Remote Notification for this user is temporarily disabled due to the multiple retry limit being exceeded or invalid DN encountered. It is re-enabled the next time the user logs in.
  - ***Off by Called Party*** indicates that a called party has temporarily disabled a user’s RN by pressing **3** on the telephone keypad.
  - ***Off due to Bad DN*** indicates that the target DN in the user’s remote notification schedule may be bad. This may happen if the target passes the restriction/permission tests but is not a real phone number or if the restriction/permission codes have been modified and the DN is now restricted where previously it was permitted.
- ***Keypad Interface*** - When this field is “Enabled”, users are able to change their schedules, periods, and targets from a telephone keypad. The default is “Enabled”.

Note: This field does not apply if Call Answering (VMUIF) is enabled. You will therefore have to create all remote notification schedules for call answering subscribers and make any requested changes once the schedules have been created.

- ***Message Remote Notification Option*** - This field determines the type of calls of which the user will be notified. Users can be notified of any calls or only those tagged as urgent. The default is “Any”.

  This field is displayed only if Voice Messaging (MMUI) is enabled. Tagged messages are a feature of voice messaging. Messages left through the call answering service can not be tagged. Since call answering subscribers do not have access to voice messaging functions, this field does not apply.

- ***Remote Notification Restriction/Permission Codes*** - The selection made in this field determines the restricted/permitted dialing codes that apply when the target DNs at which the user is to be notified are specified in the business day schedule, non-business day schedule and temporary schedule. The actual dialing codes are defined in the Voice Security Options screen, accessible from the Voice Administration menu.
- ***Busy Retry Limit -*** The number of times notification is retried at a remote phone, pager, or paging service if the destination number is busy. You may enter a value from 0 to 10.
- ***Busy Retry Interval (hh:mm) -*** This field determines how long DMS VoiceMail will wait before retrying remote notification if the destination number is busy. The valid range is from 00:00 to 23:59.
- ***No Answer Retry Limit -*** The number of times notification is retried at a remote phone, pager, or paging service if the destination number is not answered. You may enter a value from 0 to 10.
- ***No Answer Interval (hh:mm) -*** This field determines how long DMS VoiceMail will wait before retrying remote notification if the destination number is not answered. The valid range is from 00:00 to 23:59.
- ***Answered Retry Limit*** - The number of times DMS V oiceMail will retry a remote number when the number is answered but the user does not log in (by pressing “1”) or turn off further remote notification (by pressing “3”). The valid range is from 0 to 10.

  This number should be relatively low (the default is usually sufficient). If an answering machine answers the call, you do not want the RN service to keep calling back since RN can not be turned off. However, if DMS VoiceMail is calling a pager you would like the pager to go off periodically to remind the user of calls.

- ***Answered Retry Interval (hh:mm)*** - The length of time the system will wait before retrying a remote number when the destination number is answered but no messages are retrieved. The valid range is from 00:00 to 23:59.

Note: For a detailed description of retry limits and intervals refer to the section “Defining Outcalling parameters” earlier in this chapter.

- ***Business Days*** **-** This field defines business days versus non-business days. Any days for which you select “No” are considered non-business days. This information is used when creating schedules.

The following screen is displayed when you press the [Outcalling Fields] softkey.

Figure 6-8xxx
Add Local Voice User (Remote Notification Schedules)

```
  Business days Schedule:                                            MORE ABOVE
* Period 1 from (hh:mm):____ to (hh:mm):____ [Disabled] Enabled

   Target 1 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 2 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 3 DN: ___________________________  Phone  Tone Voice Numeric [Service]
                                             Pager ID:

Non-Business Days Schedule:

* Period 1 from (hh:mm):____ to (hh:mm):____ [Disabled]  Enabled

   Target 1 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 2 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 3 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:

Temporary Schedule up to midnight of (dd/mm/yy):_________________________________

* Period 1 from (hh:mm):_____ to (hh:mm):_____    [Disabled]  Enabled

   Target 1 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 2 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:
   Target 3 DN: ___________________________  [Phone] Tone Voice Numeric Service
                                             Pager Callback Number:

The Outcalling Fields data will be saved only if the user is saved.
 Return to
Basic Fields
```

* There are actually three periods listed for each schedule, each with three targets.

- ***Schedules -*** Up to three remote notification schedules can be defined for each user. One for business days, one for non-business days and a temporary schedule for short-term remote notification. (The temporary schedule overrides the Business and Non-Business days schedules until midnight of the date specified, including the current day. When the duration expires, the schedule status is automatically set to “Disabled”.)

To enable a schedule, define a valid time period and set the appropriate schedule to “Enabled” (defining the time period alone will not automatically enable the schedule). For a time period to be valid, the times must be chronologically correct, non-overlapping, within the 24-hour time window (midnight to midnight) and the targets must be dialable, non-restricted phone or pager numbers.

Within each schedule, you can define up to three time periods. For each time period, you can define up to 3 RN target DNs. The target DN can be a phone number, a directly dialable pager number, or a common pager service number (if this is a general access pager service, such as SkyPager).

For each target DN that you enter you must define the type of device to which the service will be outcalling. If the device is a phone, select “Phone”. You do not have to enter anything in the *Pager Callback Data* field. To define a pager as the target device, select one of the following options:

- ***Tone -*** to define either a Tone-only or Tone and Voice pager. You do not need to enter anything in the *Pager Callback Data* field.
- ***Voice -*** to define a Tone and Voice pager. You do not need to enter anything in the *Pager Callback Data* field.
- ***Numeric -*** to define a digital or numeric pager with DID access. Fill in the call-back number to be displayed in the *Pager Callback Data* field. If you do not enter a callback number here, the *Default Numeric Pager Data* field in the Outcalling Options screen will be displayed in which you can enter the callback number.
- ***Service -*** to define a digital or numeric pager with general access. Enter the pager’s PIN number in the *Pager ID* field. In this case, the call-back number is taken from the *Default Numeric Pager Data* field in the Outcalling Options screen (see page 8-124). This is a system-wide call-back number that is displayed on all pagers configured with “Service” as the RN target device. You may also have to change the *Numeric Pager Data Terminator* field (also in Outcalling Options). If the paging service accepts the # terminator, leave the default setting as it is. If the service does not accept this terminator, make sure this field is blank.

Note: To delete a time period, delete the associated “from” and “to” times and save the settings. To temporarily disable a time period, select “Disabled”.

If the *Keypad Interface* field is set to “Yes”, users can set create their own schedules using mailbox commands. (Note that this is not possible for Call Answering subscribers.)

For examples of retry scenarios, refer to the section “Remote Notification” earlier in this chapter.

Procedure 6-7xxx
Setting outcalling parameters for local voice users

Starting point : Add Local Voice User screen, [Change Defaults] entered.

1 Press [Outcalling Fields].

The Outcalling fields are now accessible (Figure 6-7). Use the cursor to move to the bottom of the screen and scroll upwards.

2 Move the cursor to the field to be changed or, if no change is desired, proceed to step 5.

3 Enter the data, or select an option, as required.

4 Use [Return to Basic Fields] to return to the Add Local Voice User: Change Defaults screen.

5 Go to step 5a to save any changes made to default and/or outcalling fields, or 5b to cancel all changes.

a. Use [Save].

The system saves the new user, including any changes to the default and/or outcalling fields, and prompts for another local voice user's mailbox number. To add another user, enter the next mailbox number and press <Return>. If you do not want to add another user at this time, go to step 5b.

b. Use [Cancel].

New user information is discarded, including any changes to the defaults and/or outcalling fields. The Add New User softkeys are displayed; see "Adding new users" earlier in this chapter for details.

Adding remote voice users

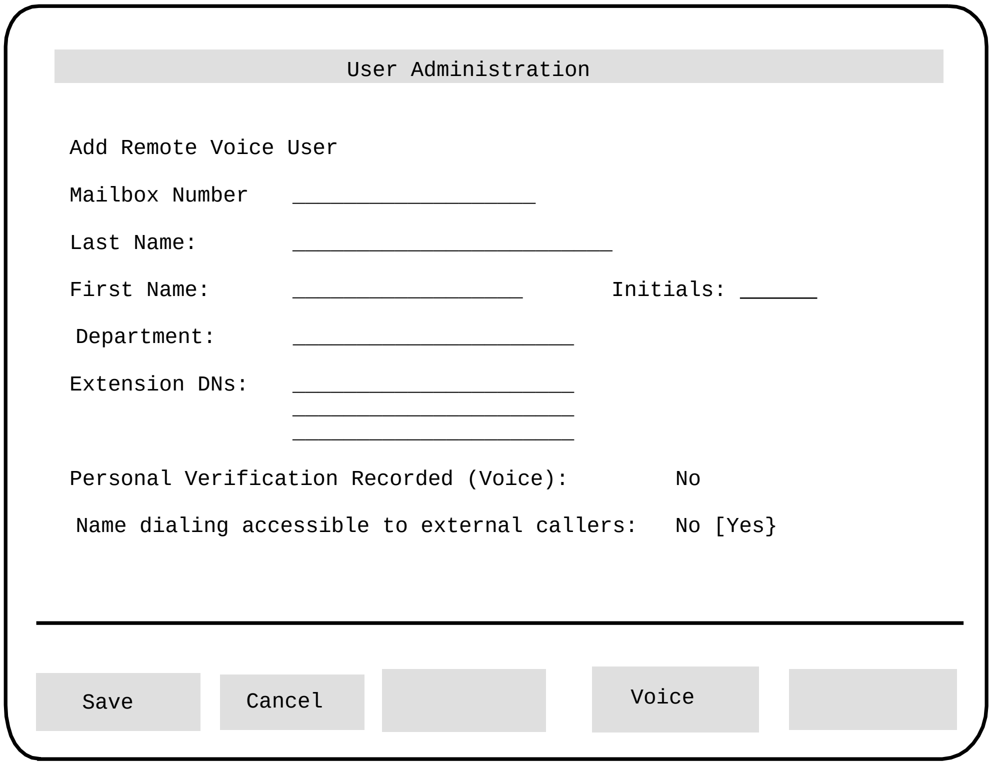
Remote voice users only exist on CPE systems that have Meridian Networking installed. The [Add Remote Voice User] softkey will not be displayed if Networking isn't installed.

Users at remote DMS Voice Mail sites that are networked to yours through Networking can be added to your system as Remote Voice Users. This is by no means necessary. There are, however, two benefits of doing this:

- When a remote voice user sends a message to a user at the local site, the sender's personal verification is played. When a user at a remote site (that is not defined as a remote voice user) sends a message, the mailbox number (e.g., 63385443, if the dialing plan is ESN) is played to the recipient.
- Remote voice users can be added to distribution lists, whereas users at remote sites (not defined as remote voice users) can not.
- Users at the local site can use name dialing to reach remote voice users.
- External callers to your system can reach remote voice users by name dialing (for example, through a voice menu or thru-dialer) if you enable the *Name Dialing Accessible to External Callers* field.

You may therefore only choose to add those users who correspond frequently with users at the local site or if it is important that they can be included in your system distribution lists. For more information about Meridian Networking, see the chapter "Meridian Networking".

Figure 6-9
Add Remote Voice User screen



The following fields are displayed:

- ***Mailbox Number*** - Enter the mailbox number preceded by any necessary access codes and the user's location prefix. The access code is the number used to dial out of the system (such as "6" for ESN). The location prefix will depend on the type of dialing plan used at the remote site. It may be an ESN prefix, a mailbox prefix, a dialing prefix or, if the remote site is part of a CDP dialing plan, the CDP steering code is already part of the mailbox number and no additional prefixes need to be added. (Dialing plans and location prefixes are described in the "Meridian Networking" chapter.) The length of the entry is limited to 18 characters.
- ***Last Name*** - The last name of the remote voice user. This field holds up to 41 characters. It accepts any characters with the exception of "+", "_", or "?". However, it is recommended that you use alphanumeric characters only. The default is blank.

Important: If you must change a user's last name once the mailbox has been added and in use, do not modify this field. Instead, make sure the user has listened to all of his or her messages, delete the mailbox and re-add it with the new last name. DMS VoiceMail uses the user's last name to keep track of users, mailboxes and messages. Modifying the *Last Name* field can cause inconsistencies.

- ***First Name*** - The first name of the new local voice user. This field can hold up to 21 characters. Spaces and hyphens (-) are allowed. The default is blank.
- ***Initials*** - The initials of the local voice user. This field can hold up to 5 alphanumeric characters. This field is for display only and can be used to distinguish users with identical first and last names. These initials, however, cannot be used during name dialing.

  Note: If you do not enter any initials, the system will automatically fill in this field with the first initial of the user's first name.

- ***Department*** - You may enter up to 31 characters. The characters "+", "?" and "_" are restricted. It is recommended that you use alphanumeric characters only and avoid using special characters altogether (even though some are accepted by this field). When adding the first user to the system, this field will be blank by default. For subsequent users, this field defaults to the department entered for the last user added.
- ***Extension DNs*** - Enter the user's full extension number at the remote site, including any necessary access codes and location prefixes. You can enter up to three DNs (the primary DN is required the others are optional). For example, if the remote voice user is part of an ESN dialing plan, the access code will likely be "6". This is followed by the ESN prefix for that site ("233") and the mailbox number "4433", making the full extension 62334433).
- ***Personal Verification Recorded (Voice***) - When a personal verification has been recorded, this field is set to "Yes". This field is only changed by the administrator using the [Voice] softkey or if the remote voice user has recorded a verification from his or her own phone.
- ***Name Dialing Accessible to External Callers*** - When this field is set to "Yes", external callers can reach the remote voice user by entering their name rather than their extension. This may occur when a caller reaches one of your thru-dialers and is prompted to enter an extension or the name of the person they want to speak to. If this feature is not enabled, the callers have to enter the remote user's DN (including access code and location prefix).

  This may not be desirable for all users as a caller who is connected to your system through a voice menu can get through to any extension as long as they know the person's name. You may therefore want to set this field to "No" for those users who have their phone calls screened by a secretary. The default is "Yes".

Procedure 6-8xxx
Adding a Remote Voice User

Starting point : User Administration screen, [Add Remote Voice User] entered.

1 The [Cancel] softkey appears; you are prompted for a remote mailbox number.

2 Go to step 2a to return, or to 2b to proceed.

a. Use [Cancel].

The Add New User softkeys are displayed; see "Adding new users" earlier in this chapter for details.

b. Enter the location code and mailbox number and press <Return>.

The Add Remote Voice User screen appears (Figure 6-9).

3 Enter the information for the new user.

4 Set *Name Dialing Accessible to External Callers*, if necessary.

5 Press the [Voice] softkey to record a personal verification recording, if one is not already recorded.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Record] softkey.

c. At the sound of the beep, speak the user's name into the telephone handset.

d. Press the [Stop] softkey to stop recording.

e. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the re-cording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

See the section, "Recording personal verifications using the [Voice] softkey" on page 6-56 for more information about the recording softkeys.

6 Go to step 6a to save the new user, or 6b to cancel the addition.

a. Use [Save].

The system saves the new user and prompts for a remote voice user's mailbox number; go to step 2 to add another user, or to 6b to exit the screen.

b. Use [Cancel].

New user information is discarded. The Add New User softkeys are dis-played; see "Adding new users" for details.

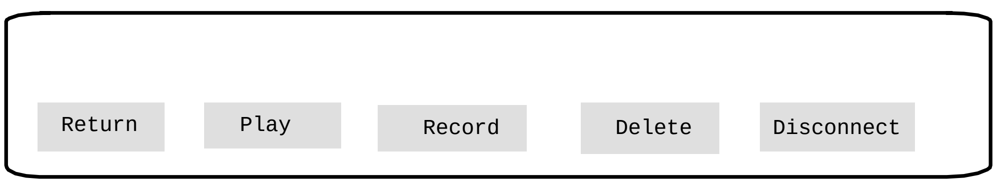
Recording Personal Verifications using the [Voice] softkey

The [Voice] softkey is used to provide a new set of softkeys for recording, playing and deleting Personal Verifications for directory entry, local and remote users. By using the voice subset of softkeys, the *Personal Verification Recorded (Voice)* field is set to “Yes” or “No”. The [Voice] softkey is available on the Add or View/Modify User screens, as well as several of the Voice Menu screens found under the Voice Administration and the Network Administration menus.

Personal verifications can only be recorded for users if Voice Messaging is installed on the system. It is not intended for Call Answering (VMUIF) subscribers. Although the administrator can still record a personal verification from this screen, it will never be used since Call Answering subscribers do not have access to the features that use the verification.

> ***Note:*** A telephone set is required to record the Personal Verification. Ensure that a phone set is available near the administration terminal where you are working.

Figure 6-10xxx
Personal Verification recording softkeys



Procedure 6-9xxx
Recording, playing, and deleting Personal Verifications

Starting point : The Main Menu.

1 Select User Administration.

2 Select one of the following: Add New User or View/Modify User.

3 If you selected Add New User, go to step 3a. If you selected View/Modify User, go to step 3b.

a. Select one of the following softkeys: [Add Directory Entry User], [Add Local Voice User], or [Add Remote Voice User]. Enter the extension number when prompted and press <Return>.

b. Select one of the following softkeys: [View/Modify Directory Entry User], [View/Modify Local Voice User], or [View/Modify Remote Voice User]. Enter the extension number when prompted and press <Return>.

4 Press the [Voice] softkey.

You are prompted for an extension number.

5 Enter the extension number of the phone set you are going to use to record a spoken name.

The phone will ring when you finish entering the extension.

6 Pick up the telephone handset.

7 To record a new verification, go to step 7a. To listen to the existing personal verification, go to step 7b. To delete the existing personal verification, go to step 7c. To return to the original set of softkeys, go to step 7d.

a. Press the [Record] softkey. At the sound of the beep speak the personal verification for the user into the handset.

When you pressed the [Record] softkey, a new [Stop] softkey appeared in its place.

Press the [Stop] softkey to stop recording.

b. Press the [Play] softkey.

If a verification has been recorded for the user recorded, it is played over the phone.

c. Press the [Delete] softkey.

If a verification has been recorded, it is deleted. A prompt is displayed advising you that the recording was deleted.

d. If you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the recording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension.

View/Modify User

Use the View/Modify User function to modify the parameters of existing DMS VoiceMail users. Selecting the View/Modify User option from the User Administration screen displays a new set of softkeys for viewing or modifying a Directory Entry User, Local Voice User, or Remote Voice User.

> ***Note:*** DMS VoiceMail supports up to four administration terminals (one main administration terminal for system administration and up to three secondary terminals for user administration only). If your system has multiple administration terminals, only the first administrator who logs on to perform user administration on a particular user can modify or delete that user. Screens will be read-only for other administrators who then access that specific user profile.

Procedure 6-10xxx
Viewing/Modifying existing user parameters

Starting point : User Administration screen, <2> entered.

1 The following softkeys appear: [View/Modify Directory Entry User], [View/Modify Local Voice User], [View/Modify Remote Voice User], and [Cancel].

2 Choose step 2a to modify a directory entry user, 2b to modify a local voice user, 2c to modify a remote voice user, or 2d to return to the User Administration screen.

a. Use [Modify Directory Entry User].

See the next section, "View/Modify Directory Entry User", for details.

b. Use [View/Modify Local Voice User].

See "View/Modify Local Voice User" later in this chapter for details.

c. Use [View/Modify Remote Voice User].

See "View/Modify Remote Voice User" later in this chapter for details.

d. Use [Cancel].

The User Administration screen and its softkeys are redisplayed.

View/Modify Directory Entry User

When you choose to view or modify a directory entry user, you are prompted for an extension number. If more than one directory entry user is associated with that extension you will see the List of Directory Entry Users screen (the top screen illustrated in Figure 6-11). From the list of users, choose the user you want to view or modify. Once you have specified the user, the View/Modify Directory Entry screen is displayed (the bottom screen illustrated in Figure 6-11). If only one user is associated with the extension you enter, the View/Modify Directory screen is displayed immediately. The fields on this screen are identical to those on the Add Directory Entry User screen, described on page 6-7.

Note: If Call Answering (VMUIF) is enabled, users can not be added as directory entry users.

Figure 6-11xxx
View/Modify Directory Entry User screen

```
                        User Administration

   View/Modify Directory Entry User

   List of Directory Entry Users whose extension number (DN)
   matches 7000

   Name                      Department
   Adams, Joan               Coordination
   Smith, John               Administration

 Move the cursor to the item and press the spacebar to select it.

 Exit    View/Modify
            User
```

```
                        User Administration

   View/Modify Directory Entry User

   Last Name:        Smith_________________________

   First Name:       John____________________     Initials: ____

   Department:       Coordination__________

   Extension DNs:    7000______________________
                     7001______________________
                     7002______________________

   Personal Verification Recorded (Voice):       [No] Yes

   Name dialing accessible to external callers:    No [Yes]

 Save    Cancel                  Voice
```

Note: If you have logged on to a terminal while another administrator is modifying the same user, only the [Exit] softkey will be displayed.

Procedure 6-11xxx
Viewing/Modifying parameters for directory entry users

Starting point : User Administration screen, [View/Modify Directory Entry User] entered.

1 The [Cancel] softkey appears; you are prompted for an extension number.

2 Go to step 2a to return or 2b to proceed.

a. Use [Cancel].

The View/Modify User softkeys appears; see “View/Modify User” earlier in this chapter for details.

b. Enter the extension number and press <Return>.

If only one user is assigned to the extension number, the View/Modify Directory Entry User screen appears (Figure 6-11).

If more than one user share the extension, the List of Directory Entry Users screen appears. Select a user by placing the cursor on the desired user, pressing <Space Bar>, and then [View/Modify User].

3 Modify the fields as needed.

4 Press the [Voice] softkey to record a personal verification recording, if one is not already recorded.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Record] softkey.

c. At the sound of the beep, speak the user’s name into the telephone handset.

d. Press the [Stop] softkey to stop recording.

e. Press the [Disconnect] softkey.

f. Press the [Return] softkey.

See the section, “Recording personal verifications using the [Voice] softkey” on page 6-56 for more information about the recording softkeys.

5 Go to step 5a to save the new user, or 5b to cancel the addition.

a. Use [Save].

The system saves the new directory entry user and prompts for another extension number. To add another user, go to step 2b. If you don’t want to add another user at this time, go to step 5b.

b. Use [Cancel].

If you have not saved the new user data, any changes will be discarded. The User Administration Menu is displayed.

View/Modify Local Voice User

Use the View/Modify Local Voice User screen to change the parameters of an existing local voice user. Any changes made to the settings affect only the specific user, not the user model. This screen contains the same fields as the Add Local Voice User - Change Defaults screen, except that Volume ID is read-only and some additional fields, shown in Figure 6-12, are displayed at the bottom of the screen.

> ***Important:*** If you must change a local voice user's last name once the mailbox has been added and in use, do not modify this field. Instead, make sure the user has listened to all of his or her messages, delete the mailbox and re-add it with the new last name. DMS VoiceMail uses the user's last name to keep track of users, mailboxes and messages. Modifying the *Last Name* field can cause inconsistencies.

Figure 6-12xxx
Additional fields in the View/Modify Local Voice User screen

```
                      User Administration                <MORE ABOVE

     View/Modify Local Voice User


     Password Last Changed    08/10/90  16:36

     Invalid Logon Attempts:  0

     Time of Last Logon:      08/21/90  09:45

   * Time of last mailbox lockout: 0/00/80   00:00

   * Calls rejected after mailbox full:   No

   **Internal Personal Greeting Recorded: [No] Yes

     External Personal Greeting Recorded: [No] Yes

   ________________________________________________________________

   Save       Cancel      Outcalling   Voice       Change
                           Fields                  Password
```

* These fields are displayed only if Call Answering (VMUIF) is enabled. **This field is displayed only if Voice Messaging (MMUI) is enabled.

Note: If you have logged on to a terminal while another administrator is modifying the same user, only the [Exit] softkey will be displayed.

See "Adding local voice users" and "Changing defaults for local voice users" earlier in this chapter, for field descriptions.

The additional fields in the View/Modify Local Voice User screen are:

- ***Password Last Changed*** - This is a read-only field displaying the date and time of the last password change. For new Voice Messaging users, this is the time at which the user was added. For Call Answering subscribers, the time is set to “nil”.

  If Voice Messaging (MMUI) is installed on the system, there is a maximum imposed on the number of days permitted between password changes. This value is set in the Voice Security Options screen. If this maximum is exceeded, the user’s mailbox is disabled. To re-enable a disabled mailbox, set the *Logon Status* field to “Enabled”. You should also ensure that the user understands why the mailbox was disabled and confirm that he or she is aware of the password expiry limit.

- ***Invalid Logon Attempts*** - This is a read-only field displaying the number of successive logon attempts using an incorrect password. When the maximum number of invalid logon attempts is reached, the user’s mailbox is disabled.

  A large number of invalid logon attempts may indicate a security problem. For example, someone may be trying to get into your system through this particular mailbox. Should this value be suspiciously high, contact the owner of the mailbox and determine if he or she has had problems logging in. The owner may have simply forgotten the mailbox password and tried a variety of passwords. If you are sure that there is no security risk, re-enable the mailbox by setting the *Logon Status* field to “Enabled”. This action resets the *Invalid Logon Attempts* field to “0”.

- ***Time of Last Logon*** - This is a read-only field displaying the time of the last successful logon. In the case of a new user who has not logged on yet, no date or time will be displayed.

  A considerable amount of time between the current date and the user’s last logon could indicate one of several things. For private users: the user may be on holiday or off-site and not retrieving messages; the user may have left the organization. For both CO and CPE environments: the user may not know how to log on and retrieve messages; the user may have forgotten his or her password (in which case he or she may have stopped trying to log on and has not contacted the administrator to change the mailbox password). Try to contact the user to determine if there is a problem. You might also want to check the voice messaging user usage report (described in the “Operational Measurements” chapter) to see if the user has messages waiting.

- ***Time of last mailbox lockout*** - This field is only applicable if Call Answering (VMUIF) is enabled. It is a read-only field displaying the time of the last mailbox lockout. This is usually due to an excessive number of invalid logon attempts. To re-enable a disabled mailbox, set the *Mailbox Status* field to “Enabled”.

- ***Calls rejected after mailbox full* -** This field is only applicable if Call Answering (VMUIF) is enabled. If any calls have been rejected due to a full mailbox, this field will display “Yes”. “No” either indicates that the mailbox is not full or that the subscriber’s mailbox is full but no calls have been rejected.

  You may never actually see this field set to “Yes” because when the user logs on, this field is reset to “No”. When a user logs on after messages have been lost, he or she will hear a message indicating that the mailbox is full and that messages have been lost. In turn, the user may inform you of lost messages. Ask the user to delete messages if this has not already been done. If both you and the user agree that a larger mailbox is needed, you can increase the value in the *Storage Limit* field in the View/Modify Local Voice user screen.

- ***Internal Personal Greeting Recorded* -** This is a read-only field. It does not apply if the VMUIF interface is installed. (In this case, only external greetings can be recorded.) If the interface is MMUI, it indicates whether or not an internal personal greeting has been recorded by the user. This greeting is played to callers that have reached the user from a line inside the switch.

  This greeting may be less formal and can include information that is not appropriate to external callers. For example, *“Hi, this is David. I’m not at my desk right now, so please leave a message after the tone. If this is an urgent matter, you can find me at Brian’s desk.”*

- ***External Personal Greeting Recorded*** - This is a read-only field. It indicates whether or not an external personal greeting has been recorded by the user. For private (business) users, this greeting is played to callers who reach the user’s mailbox from an outside trunk. This message should be more formal than the internal greeting. For Call Answering subscribers, this is the greeting played to all callers who reach the subscriber’s mailbox.

Procedure 6-12xxx
Viewing/Modifying parameters for local voice users

Starting point : User Administration screen, [View/Modify Local Voice User] entered.

1 The [Cancel] softkey appears; you are prompted to enter a mailbox number.

2 Go to step 2a to return, or 2b to proceed.

a. Use [Cancel].

The Modify User softkeys appear; see “View/Modify User” earlier in this chapter for details.

b. Enter the mailbox number and press <Return>.

The View/modify local voice user screen appears

3 Change the fields as required.

4 If one of the fields, *Remote Notification Capability* or *Delivery to Non-Users* is enabled, then use [Outcalling Fields] to display the *Outcalling* fields for editing. See “Defining Outcalling parameters” for descriptions of the fields.

5 If a personal verification has not been recorded for this user, press the [Voice] softkey.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Play] softkey to see if a verification has been recorded.

If there is no verification, or if you want to record a new one, continue with step 5c. If you do not need to re-record the verification, go to step 5f.

c. Press the [Record] softkey.

d. At the sound of the beep, speak the user’s name into the telephone handset.

e. Press the [Stop] softkey to stop recording.

f. Press the [Disconnect] softkey.

g. Press the [Return] softkey.

See the section, “Recording personal verifications using the [Voice] softkey” on page 6-56 for more information about the recording softkeys.

6 Use [Change User Password] to set the user’s DMS VoiceMail password.

You are prompted to enter the new password, then to re-enter the new password to verify it. The passwords are not displayed on the screen.

User passwords must be numeric and up to 16 digits long. By default, the initial password for a new user is the same as the user’s mailbox number.

7 Go to step 7a to save the new user, or 7b to cancel the addition.

a. Use [Save].

The system saves the new user and prompts for another local voice user’s mailbox number. To view or modify another user, go to step 2b. If you do not want to modify another user at this time, go to step 7b.

b. Use [Cancel].

New user information is discarded. The User Administration menu appears with the modify user softkeys displayed at the bottom.

View/Modify Remote Voice User

> ***Note:*** Remote voice users can only be added to CPE systems that have Meridian Networking.

If you have added remote voice users to your system, you can modify them at this screen (see Figure 6-13). The fields displayed on the View/Modify Remote Voice User screen are identical to those on the Add Remote Voice User screen and are described on page 6-53.

> ***Important:*** If you must change a remote voice user's last name once the mailbox has been added and in use, do not modify this field. Instead, make sure the user has listened to all of his or her messages, delete the mailbox and re-add it with the new last name. DMS VoiceMail uses the user's last name to keep track of users, mailboxes and messages. Modifying the *Last Name* field can cause inconsistencies.

Figure 6-13xxx
View/Modify Remote Voice User screen

```
                         User Administration
View/Modify Remote Voice User

Mailbox Number:     2226000

Last Name:         Jones

First Name:        Ed                      Initials: ____

Department:        Accounting

Extension DNs:     6000
                   _______________
                   _______________

Personal Verification Recorded (Voice)         [No]  Yes

Name Dialing Accessible to External Callers:    No  [Yes]
________________________________________________________________

  Save        Cancel                     Voice
```

```
Note: If you have logged on to a terminal while another administrator
      is modifying the same user, only the [Exit] softkey will be
      displayed.
```

Procedure 6-13xxx
Viewing/Modifying parameters for Remote Voice Users

Starting point : User Administration screen, [View/Modify Remote Voice User] entered.

1 The [Cancel] softkey appears; you are prompted for a remote mailbox number.

2 Go to step 2a to return, or 2b to proceed.

a. Use [Cancel].

View/Modify User softkeys appear; see "View/Modify User" earlier in this chapter.

b. Enter the location code, mailbox number, and press <Return>.

The View/Modify Remote Voice User screen appears (Figure 6-13).

3 Make the required changes.

4 If a personal verification has not been recorded for this user, press the [Voice] softkey.

a. Enter the extension number of the phone you will be using to record the verification.

A new set of softkeys is displayed.

b. Press the [Play] softkey to see if a verification has been recorded.

If there is no verification, or if you want to record a new one, continue with step 5c. If you do not need to re-record the verification, go to step 5f.

c. Press the [Record] softkey.

d. At the sound of the beep, speak the user's name into the telephone handset.

e. Press the [Stop] softkey to stop recording.

f. Press the [Disconnect] softkey.

g. Press the [Return] softkey.

See the section, "Recording personal verifications using the [Voice] softkey" on page 6-56 for more information about the recording softkeys.

5 Go to step 5a to save the new user, or 5b to cancel the addition.

a. Use [Save].

The system saves the new user information. You are prompted for another mailbox number. To modify another remote voice user go to step 2b. If you do not need to modify any other users, go to step 5b.

b. Use [Cancel].

If you have not saved the user data, any changes that you have made are discarded. The User Administration appears with the modify user softkeys displayed at the bottom.

Delete User

Use this function to delete existing DMS VoiceMail users. When you select the Delete User item from the User Administration screen, a new set of softkeys is displayed (see Figure 6-14) with which you can specify whether you wish to delete a Directory Entry User, Local Voice User, or Remote Voice User. The Delete User screens are similar to the View/Modify User screens, however, fields on the Delete User screens are read-only and user data cannot be modified. For field descriptions, see the section "View/Modify User" on page 6-58.

Figure 6-14xxx
Delete User softkeys

```
                    User Administration

      1 Add New User
      2 View/Modify User
      3 Delete User
      4 Find Users
      5 View/Modify User Models
      6 Distribution Lists

   Select an item >

            Cancel     Delete        Delete       Delete
                       Directory     Local        Remote Voice
                       Entry User*   Voice User   User**
```

* This softkey is displayed only if Voice Messaging (MMUI) is enabled.
**This softkey is displayed only if Meridian Networking is enabled.

Before deleting a user, you may want to ensure that there are no voice messages in the user's mailbox. This can be verified by checking the *Storage Used* field in the Modify Local Voice User screen. If there are messages remaining, you may want to make sure that the user listens to them before you delete the user.

CAUTION
Deleting mailboxes
If a user’s mailbox is removed before user usage data is processed then the data will be lost. (See the chapter “Operational Measurements”.) To avoid this situation, do not delete the mailbox until the data is processed. (If you have the AdminPlus feature and file downloading capability then do not delete the mailbox until you have downloaded the data.) Instead, the mailbox should be disabled. See the description of the *Logon Status* field in the “View/Modify Local Voice User” section earlier in this chapter. Once data is processed then you can delete the user if you wish.

Procedure 6-14xxx
Deleting users

Starting point : User Administration screen, <3> entered.

1 The following softkeys appear: [Delete Directory Entry User], [Delete Local Voice User], [Delete Remote Voice User], and [Cancel].

2 Choose step 2a to delete a directory entry user, 2b to delete a local voice user, 2c to delete a remote voice user, or 2d to return to the User Administration screen.

a. Use [Delete Directory Entry User].

See “Deleting a Directory Entry User” earlier in this chapter for details.

b. Use [Delete Local Voice User].

See “Deleting a Local Voice User” earlier in this chapter for details.

c. Use [Delete Remote Voice User].

See “Deleting a Remote Voice User” earlier in this chapter for details.

d. Use [Cancel].

The User Administration screen and its softkeys are redisplayed.

Deleting a Directory Entry User

When you gain access to the Delete Directory Entry User screen, you are prompted to enter the extension number of the user. The first screen in Figure 6-15 is displayed if more than one user is associated with the extension. The second screen is displayed if only one user has that extension. Note that when you delete a directory entry user, their personal verification is automatically deleted.

> ***Note:*** This does not apply to Call Answering subscribers since they can not be added to the system as directory entry users.

Figure 6-15xxx
Delete Directory Entry User screen

```
                    User Administration

  Delete Directory Entry User

  List of Directory Entry Users whose extension number (DN)
  matches 7000

  Name                    Department
  Adams, Joan             Coordination
  Smith, John             Administration

Move the cursor to the item and press the spacebar to select it.

  Exit         Delete
               User
```

```
                    User Administration

  Delete Directory Entry User

  Last Name:         Smith

  First Name:        John              Initials:

  Department:        Administration

  Extension DNs:     7000
                     7001
                     7002

  Personal Verification Recorded (Voice):        Yes

  Name dialing accessible to external callers:   No   Yes

  OK to
  Delete
```

Note: If you have logged on to a terminal while another administrator is modifying the specified user, only the [Exit] softkey will be displayed.

Procedure 6-15xxx
Deleting directory entry users

Starting point : User Administration screen, [Delete Directory Entry User] entered.

1 The [Cancel] softkey appears; you are prompted for an extension number.

2 Go to step 2a to return, or 2b to proceed.

a. Use [Cancel].

The Delete User softkeys are displayed; see “Delete User” on page 6-68 for details.

b. Enter the extension number and <Return>.

If only one user is assigned to the extension number, the Delete Directory Entry User screen appears with the cursor positioned at Last Name (Figure 6-15). Proceed to step 3.

If more than one user shares the extension number, the List Directory Entry Users screen appears. Select the required user and press [Delete User]. The Delete Directory Entry User screen appears; proceed to step 3.

If you do not wish to proceed, use [Exit] and go to “Delete User”.

3 Choose step 3a to delete the user, or 3b to cancel.

a. Use [OK to Delete].

The user is deleted and the system prompts for an extension number; go to step 2 to delete another user, or to 3b to leave the menu.

b. Use [Cancel].

The deletion is canceled. The Delete User softkeys appear; see “Delete User” for details.

Deleting a Local Voice User

Local voice users can be removed from the system at the Delete Local Voice User screen (Figure 6-16). When you delete a local voice user, the user's mailbox (including all messages), Personal Verification, any personal greetings, and all entries of that user in system distribution lists are deleted.

Figure 6-16xxx
Delete Local Voice User screen

```
                        User Administration
   Delete Local Voice User

   Mailbox Number:     7000      Volume ID: 203

   Last Name:          Smith

   First Name:         John             Initials:

 * Department:         Administration

   Extension DNs:      7000
                       7001
                       7002

   Revert DN:          0

   Message Waiting Indication Options:  Any

   Message Waiting Indication DN:       8765432

   Message Waiting Link Name:           Link1

   Personal Verification Recorded (Voice)         Yes

 * Name Dialing Accessible to External Callers:    Yes
 ________________________________________________________ MORE BELOW

  OK to       Cancel      Outcalling
  Delete                  Fields
```

* These fields are displayed only if Voice Messaging (MMUI) is enabled.

Note: If you have logged on to a terminal while another administrator is modifying the same user, only the [Exit] softkey will be displayed.

Procedure 6-16xxx
Deleting local voice users

Starting point : User Administration screen, [Delete Local Voice User] entered.

The [Cancel] softkey appears; you are prompted to enter a mailbox number.

1 Go to step 1a if you do not want to delete the user. Go to step 1b to proceed.

a. Use [Cancel].

The Delete User softkeys are displayed.

b. Enter the extension number of the user you want to delete and press <Return>.

The Delete Local Voice User screen is displayed.

2 Choose step 2a to delete the user, or 2b to cancel.

a. Use [OK to Delete].

The user is deleted and the system prompts for another extension number. To delete another user go to step 1b. If you do not want to delete another user at this time, go to step 2b.

b. Use [Cancel].

The user is not deleted. The Delete User softkeys are displayed.

Deleting a Remote Voice User

Remote voice users can be removed from your system from the Delete Remote Voice User screen (Figure 6-17). When you delete a remote voice user all references to the remote user in the directory are removed along with their personal verification (if recorded).

Figure 6-17xxx
Delete Remote Voice User screen

```
                      User Administration

 Delete Remote Voice User

 Mailbox Number:     22600
 Last Name:          Jones
 First Name:         Ed                Initials:
 Department:         Accounting
 Extension DNs:      6000

 Revert DN:       0
 Personal Verification Recorded (Voice)         Yes
 Name Dialing Accessible to External Callers:   Yes
                                                        MORE BELOW

 OK to       Cancel
 Delete
```

Note: If you have logged on to a terminal while another administrator is modifying the specified user, only the [Exit] softkey will be displayed.

Procedure 6-17xxx
Deleting remote voice users

Starting point : User Administration screen, [Delete Remote Voice User] entered.

1 The softkey set changes to display [Cancel], and you are prompted to enter a remote mailbox number.

2 Go to step 2a to return, or 2b to proceed.

a. Use [Cancel].

The Delete User softkeys are re-displayed.

b. Enter the location code and mailbox number, and press <Return>.

The Delete Remote Voice User screen appears (Figure 6-17).

3 Choose step 3a to delete the user, or 3b to cancel.

a. Use [OK to Delete].

The user is deleted and the system prompts for a remote voice user's mailbox number; go to step 2 to delete another user, or to 3b to leave the menu.

b. Use [Cancel].

The deletion is canceled. The Delete User softkeys appear.

Find Users

Find Users, item <4> in the User Administration menu, allows you to search for and view existing users or print a list of existing users, therefore serving as a record-keeping tool. You need only complete fields to the extent necessary to identify a user or group of users. Once you have filled in the Find Users screen, you may either view users on the screen or print a list of users. If you are searching for a specific user in order to modify that user's configuration, select the [View Users] softkey. When you find the user you are searching for, you may choose to modify that user from the List of Users screen (Figure 6-19). You do not have to return to the User Administration menu.

The fields on the Find Users screen (Figure 6-18) accept three wildcard characters: "+" (the plus sign), "_" (underscore), and "?" (question mark).

The plus sign (+) is used to match a number of characters. For example, if you enter **2+** in the *Mailbox Number* field, all mailboxes beginning with 2 will be retrieved.

The underscore (_) matches a single character. For example, if you enter **210_** in the *Mailbox Number* field, mailboxes with numbers in the range 2100 to 2109 will be retrieved. To retrieve all mailboxes numbered 2100 to 2199, enter **21_ _**.

The question mark (?) produces a "sound match". This is useful if you are unsure of the spelling of a user's name. For example, a user calls to inform you that his mailbox has been disabled and tells you that his name is "John Crowe". You forget to ask him for the spelling of his last name (it could be spelled Crow or Crowe). If you enter Crow+, the system will only find all surnames that begin with Crow. If you enter Crow_, the system will find surnames that begin with Crow and are followed by one letter. If you enter Crow?, the system will find all names that sound like "Crow".

> ***Note:*** The search criteria that you specify in this screen also apply when you use the [Print Users] softkey.

Figure 6-18xxx
Find Users screen

```
                          User Administration

   Find Users

   User Type:  Any Directory_Entry_User [Local_Voice_User] Remote_Voice_User

*  Status:     [Any]  Enabled   Disabled  Expired   Violation

** Mailbox Number:    ______________________  * Volume ID: ____

   Last Name:         ___________________________________

   First Name:      ______________

   Department:      ______________________________

   Extension Number (DN):  _________________

   Personal Verification Status:   [Any]  Not_Recorded   Recorded



 ______________________________________________________________________
    Select a softkey >

  [          ]  [  Cancel  ]  [View Users]  [Print Users]  [          ]
```

```
* These fields are displayed only if the user type is Local Voice User.
**This field is displayed if the user type is either Local Voice User or
  Remote Voice User.
```

For most of the fields on the screen, see the section "Adding local voice users" for descriptions. The following fields are unique to this screen or require additional explanation:

- ***User Type*** - This field allows you to specify the type of user you want to view. Your choices are "Any", "Directory Entry User", "Local Voice User", or "Remote Voice User". The default is "Any".
- ***Status*** - This field is displayed only if *User Type* is set to "Local Voice User". It allows you to retrieve and view local users according to their mailbox status. You have five choices:
  - *Any* - Select this option if the mailbox status is not a search criterion.
  - *Enabled* - Select this option if you want to find users whose mailboxes are enabled.

- *Disabled* - Select this option to find users whose mailboxes are disabled. These users cannot log on, however messages are still received. A mailbox may be disabled if the user has made too many logon attempts with an incorrect password or if their password has expired.
- *Expired* - Select this option to find users whose passwords have expired. This situation can occur only if users are required to change their password before the number of days stipulated in the field *Maximum Days Permitted Between Password Changes* in the V oice Security Options screen. If this field is set to “0”, users passwords will never expire. If a user’s password has expired, their mailbox will be disabled and they will not be able to log on.
- *Violation* - Select this option to find users who have surpassed the maximum number of allowed invalid logon attempts for their mailbox (configured in the Voice Security Options screen). Users who have made too many invalid logon attempts will not be able to log on and their mailbox will be disabled.

- ***Mailbox Number*** - This field is displayed if *User Type* is set to “Local Voice User” or “Remote Voice User”. This field can hold up to 18 characters.
- ***Volume ID*** - This field is displayed only if *User Type* is set to “Local Voice User”. It specifies the hard disk volume to which a user is assigned. All users must be assigned to a volume.

  Information on disk usage can be obtained through the Disk Usage report (see “Traffic Reports” in the “Operational Measurements” chapter). If you notice that one volume is getting full, you should move some of the users to another volume. Set the *Volume ID* field to the ID of the volume that is almost full in order to get a list of user’s names and their mailbox numbers. You can then move some of these users to another volume with the Move User utility accessible through the Tools menu (see Appendix A)
- ***Personal Verification Status*** - You may view users according to whether or not they have a personal verification recorded. If you want to ensure that all users have a recorded personal verification, you can generate a list of of users who don’t have a recorded verification. You can then record verifications for these users or contact them and ask them to do this themselves. The default is “Any”, meaning that the personal verification status will not be used as a search criterion.

Procedure 6-18xxx
Searching for users

Starting point : User Administration screen, <4> entered.

1 The Find Users screen appears (Figure 6-18).

2 Complete the screen with the required search parameters.

3 Choose 3a to display the results, 3b to print, or 3c to cancel.

a. Use [View Users] to display search results on the screen.

See the next section, "Viewing a list of users", for details.

b. Use [Print Users] to send search results to the printer.

See "Print Users" on page 6-83 for details.

c. Use [Cancel] to cancel the search.

The User Administration screen is redisplayed.

Viewing a list of users

The List of Users screen (Figure 6-19) appears when the [View Users] softkey on the Find Users screen is used. It provides a list of user names and mailboxes matching the search parameters entered in the Find Users screen. Users are sorted by the first search parameter that is filled in on the Find Users screen.

Figure 6-19xxx
List of Users screen

```
                          User Administration

List of Users

Name         Mailbox   Department *   User Retain  Storage     Personal
                                      Type ReadMsg Used Limit Verific.
                                           (days)  (minutes)  Recorded

Alcott,Tom  2209      Financial      Loc     7       12   42    No
Gordon,John 2145      Sales          Loc     7       20   30    Yes
Jones,Tracy 2134      Admin          Loc     7        7   99    No
Smith,Bod    2291     Accounting     Loc     0       22   40    Yes
Valdez,J     212026   Marketing      Rem                        Yes

Move the cursor to the item and press the spacebar to select it.

    Exit      View/Modify   Delete User      Voice        **Next
                 User                                      Page
```

* The Department column only appears if Voice Messaging (MMUI) is enabled.
**Next Page only appears if the information fills more than one screen.

For every user on the list, the List of Users screen displays the following information:

- ***Name*** - The user's last name followed by the first name. This field can display up to 15 characters.
- ***Mailbox Number*** - The user's mailbox number. This field is empty if you are retrieving directory entry users. For local voice users, up to eight digits are displayed. For remote voice users in networked systems, the location prefix is displayed together with the mailbox number, up to 18 digits.
- ***Department*** - The user's department name. This column can hold up to 10 characters.

- ***User Type*** - This field displays an abbreviation of the type of user: “Dir” (directory entry user), “Loc” (local voice user), or “Rem” (remote voice user).
- ***Retain Read Msg (days)*** - The number of days that read messages are allowed to be retained in this user’s mailbox. A number is displayed only if the user is a local voice user.
- ***Storage Used*** - The minutes of storage used by the user, if a local voice user.
- ***Storage Limit*** - The number of minutes of storage allocated to the user, if a local voice user.
- ***Personal Verification Recorded*** - Indicates whether or not a spoken name has been recorded for this user.

Procedure 6-19xxx
Listing users matching search criteria

Starting point : Find Users screen, user parameters entered, [View Users] selected.

1 A list of all the DMS VoiceMail users matching the search parameters appears (Figure 6-19).

2 Use [Next Page] to view the next page of the list.

When [Next Page] no longer appears, the end of the list has been reached.

3 Choose step 3a to modify a user, 3b to delete a user, or 3c to return.

a. Move the cursor to the required user, press <Space Bar> to select the item, and use [View/Modify User].

Depending on the setting in the User Type field, the View/Modify Directory Entry User, View/Modify Remote Voice User, or View/Modify Local Voice User screen will be displayed. For details, see the appropriate section earlier in this chapter:

b. Move the cursor to the required user, press <Space Bar> to select it and use [Delete User].

c. Use [Exit] (at any time) to exit from the List of All Users screen.

The Find Users screen is redisplayed.

Print Users

Instead of viewing users on screen, you can obtain a printout of users by using the [Print Users] softkey. However, only the following parameters are included in the printout: Name, Mailbox Number, Department, User Type, Retain Read Msg, Storage Used, Storage Limit, and Personal Verification Recorded. To view a complete user profile with all parameters included, choose the [View/Modify] softkey from the User Administration screen.

To print a selection of users (for example, all users with mailboxes in the range 2000 to 2299), fill out the search criteria in the Find Users screen as described on page 6-77.

Figure 6-20xxx
Print Users output

8/27/92 ABC Company Page 1

List of Users

Name	Mailbox	Dept.*	User Type	Retain ReadMsg (days)	Storage Used (minutes)	Storage Limit (minutes)	Personal Verific. Recorded
Alcott, Tom	2209	Finan	Loc	7	42	12	No
Gordon, John	2145	Sales	Loc	7	20	30	Yes
Jones, Tracy	2134	Admin	Loc	7	7	99	No
Smith, Bod	2291	Accou	Loc	0	22	40	Yes
Valdez, J	212026	Marke	Rem				Yes

* The Department column only appears if Voice Messaging (MMUI) is enabled.

Procedure 6-20xxx
Printing a list of users

Starting point : Find Users screen, user parameters entered.

1 Use [Print Users].

The following softkeys appear: [Continue Printing] and [Cancel Printing].

You are prompted to make sure your printer is ready and on-line.

2 Choose step 2a to print or 2b to cancel.

a. Use [Continue Printing] to start printing.

Once printing is complete, the Find Users screen and its softkeys are redisplayed; you may stop printing at any time by proceeding to 2b.

b. Use [Cancel Printing] at any time to cancel printing.

As a result of print buffering, you may experience some delay before control is returned to your screen and the printer actually stops printing.

User Models

A user model is like a template which is configured to represent the needs of certain type of user (such as a secretary or an executive in a CPE environment, or a dial pulse user versus a deluxe DTMF user in a CO environment). For example, you would probably want to give executives a larger message storage limit than regular users.

Before you begin adding users, it is recommended that you identify categories of users and come up with a typical profile for each. For each category that you come up with, create a user model. You can create up to 15 different models.

When you are ready to add users, add them according to category if possible. For example, add all secretaries first. Set the user model to the one you have created for secretaries. This will save on the amount of time you spend on each user since most parameters will be the same. You can of course, customize a user profile and deviate from the user model.

Three pre-named user models are provided. If Voice Messaging is installed on the system, the three pre-named models are: “Standard”, “Executive” and “Secretary”. If Call Answering is installed on the system, the three pre-named models are: “DTMF”, “DTMF Deluxe” and “Dial Pulse”.

Feature	User Models
Voice Messaging	Standard Executive Secretary
Call Answering	DTMF DTMF Deluxe Dial Pulse

Table 6-3 shows the default configuration for the three pre-named Call Answering user models. The remaining un-named user models are configured the same as the DTMF model.

Table 6-3
Call Answering User Model configurations

Feature	DTMF	Dial Pulse	DTMF Deluxe
Storage limit (minutes)	10	10	15
Message Waiting Indication Option	Any	Any	Any
Logon Status	Enabled	Enabled	Enabled
Lockout Duration	00:00	00:00	00:00
Billing Class	0	0	0
Read Message Retention (days)	3	2	4
Dial Pulse Support	No	Yes	No
Auto Logon	Yes	Yes	Yes
Delayed Prompts	Yes	Yes	Yes
Skip to First New Message	No	No	No
Callers Notified of Busy Line	Yes	Yes	Yes
Custom Revert restriction/permission codes	Local	Local	Local

The primary differences between these models are in the storage limit (the deluxe model has the greatest value), the read message retention (the deluxe model again has the greatest value), and dial pulse support (Dial Pulse model only).

Table 6-4 shows the default configuration for the three pre-named Voice Messaging user models. The remaining un-named user models are configured the same as the Standard model.

Table 6-4
Voice Messaging User Model configurations

Feature	Standard	Executive	Secretary
Storage limit (minutes)	4	20	10
Message Waiting Indication Option	Any	Any	Any
Name Dialing Accessible to External Callers	Yes	Yes	Yes
Administrator Capability	No	No	No
Logon Status	Enabled	Enabled	Enabled
Billing Class	0	0	0
Read Message Retention (days)	0	0	0
Retain Copy of Sent Messages	Yes	Yes	Yes
Auto Logon	No	No	No
Delayed Prompts	Yes	Yes	Yes
Auto Play	No	No	No
Callers Notified of Busy Line	Yes	Yes	Yes
Custom Revert Restriction/ Permission codes	Local	Local	Local
Extension Dialing restriction/permission codes	Local	Local	Local
External Call-Sender restriction/permission codes	Local	Local	Local

The only difference between the default configurations of the three pre-named models is in the storage limit (the Secretary model allows for more storage than the Standard model and the Executive model allows for the most storage.)

These default values and model names are only meant as a guideline and a starting point. It will be up to you to configure as many user models as is necessary to meet the various requirements of the different types of users that will be added to your system. It is therefore recommended that before you configure your user models, you perform an analysis of user types and needs. Having well-defined user models based on real user needs will make the process of adding users to the system much easier.

Outcalling features (Delivery to Non-User and Remote Notification) are disabled for all of the default user models. If you want to enable one or both of these features for any of the user models you create, you must first ensure that Outcalling is installed on the system. (This can be verified in the General Options screen.) If Outcalling is installed, you can access the Outcalling Fields in the user models and enable Delivery to Non-Users and Remote Notification for those user types that require these outcalling capabilities. (Note that only Remote Notification is available to Call Answering subscribers.)

User Models List screen

The User Models List screen shows a listing of the models that are defined in the system and the characteristics of those models. Up to 15 models can be defined. Figure 6-21 shows the three models which are pre-named when your system is installed. The User Models List screen is accessed from the User Administration screen. You can view or modify a model by selecting it on the User Models List screen and then using the [View/Modify Model] softkey.

Because User Models are not visible in the Add Local Voice User screen unless they are given a name, ensure that each user model you plan to use has a name. Conversely, if you prefer a shorter list of user models, blank the names of the superfluous models to remove them from the Add Local Voice User screen. The User Models List screen (Figure 6-21) shows all 15 user models (with and without names).

Figure 6-21xxx
User Models List screen

User Administration

User Models List

Name	Storage (mins)	Retain ReadMsg	Auto Logon	Retain Copy	Billing Class	Auto Play	Msg Waiting	Out Calling*	Preferred Language**
Standard	4	0	No	Yes	0	No	Any	No	AmericanE
Executive	20	0	No	Yes	0	No	Any	No	AmericanE
Secretary	10	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE
	4	0	No	Yes	0	No	Any	No	AmericanE

Move the cursor to the model and press the spacebar to select it.

Exit	View/Modify Model			

* This column is displayed if Outcalling is installed on your system.
**This column is displayed if you have a multilingual system.

This screen displays some of the current settings for the existing user models. These models can be modified by selecting the [View/Modify Model] softkey.

Procedure 6-21xxx
Listing user models

Starting point : User Administration screen, <5> entered.

1 The User Models List screen appears (Figure 6-21).

2 Choose step 2a to select a user model or 2b to return.

a. Select a model to view or modify; move the cursor to the model and press the [View/Modify Model] softkey.

See the following section, “View/Modify User Models”, for details.

b. Use [Exit].

The User Administration screen is re-displayed.

View/Modify User Models

The View/Modify User Models screen (Figure 6-22) is used to view or modify the user models. It is accessed from the User Models List screen using the [View/Modify Model] softkey.

Figure 6-22xxx
View/Modify User Models screen for the MMUI interface

```
                          User Administration

  View/Modify User Models

  Model Name:                            Executive

  Storage Limit (minutes):               20

  Message Waiting Indication Option:     None [Any]  Urgent

  Name Dialing Accessible to External   No   [Yes]
  Callers:

  Administrator Capability:              [No] Yes

  Logon Status:                          Disabled [Enabled]

  Billing Class                          2

  Read Message Retention (days):         0
  ("0" implies that read messages
  are retained until the user
  deletes them manually.)
  (Subject to the organization's
  maximum retention of 7 days.)

  Retain copy of Sent Messages:          [No] Yes

  Auto Logon:                            [No] Yes

  Delayed Prompts:                       No   [Yes]

  Auto Play:                             [No] Yes

  Callers Notified of Busy Line:         No   [Yes]

                                                   MORE BELOW
```

Figure continued on next page

Figure 6-22xxx
View/Modify User Models screen for the MMUI interface (continued)

```
                          User Administration                MORE ABOVE


View/Modify User Models
* Dual Language Prompting:         Disabled  [Enabled]

* Preferred Language:              [AmericanEnglish]
                                   EuropeanEnglish
                                   Mandarin

  Custom Revert                    None  On_Switch [Local] Long_distance_1
  restriction/permission codes:    Long_distance_2

  Extension Dialing                None  On_Switch [Local] Long_distance_1
  restriction/permission codes:    Long_distance_2

  External Call Sender             None  On_Switch [Local] Long_distance_1
  restriction/permission codes:    Long_distance_2

**Receive AMIS messages:           [No]  Yes

**Compose/send AMIS messages:      [No]  Yes

**AMIS restrict'n/permission codesNone  On_Switch [Local] Long_distance_1
                                   Long_distance_2

  Save        Cancel      Outcalling
                          Fields
```

* These fields are only displayed if more than one language is installed.
**These fields are only displayed if AMIS networking is enabled.

Note: If you log on while another administrator is modifying the same user model, the fields will be read-only and only the [Exit] soft key will be displayed.

Figure 6-23xxx
View/Modify User Models screen for the VMUIF interface

```
                         User Administration

View/Modify User Models
Model Name:                         DTMF Deluxe
Storage Limit (minutes):            20
Message Waiting Indication Option:  None [Any]  Urgent

Logon Status:                       Disabled [Enabled]
Lockout Duration (hh:mm):           00:00
(00:00 implies no mailbox reset)
Billing Class                       2
Read Message Retention (days):      10
("0" implies that read messages
are retained until the user
deletes them manually.)
(Subject to the organization's
maximum retention of 7 days.)
                                                       MORE BELOW
```

```
                         User Administration               MORE ABOVE

View/Modify User Models
   Dial Pulse Support:              [No] Yes
   Auto Logon:                      [No] Yes
   Delayed Prompts:                 No[Yes]
   Skip to First New Message:       [No] Yes
   Callers Notified of Busy Line:   No[Yes]
*  Preferred Language:              [AmericanEnglish]
                                    EuropeanEnglish
                                    Mandarin
   Custom Revert                    None  On_Switch [Local] Long_distance_1
   restriction/permission codes:    Long_distance_2
**Receive AMIS messages:            [No] Yes

   Save        Cancel      Outcalling
                             Fields
```

* This field is displayed only on multilingual systems.
**This field is displayed only if AMIS networking is enabled.

Note: If you log on while another administrator is modifying the same user model, the fields will be read-only and only the [Exit] soft key will be displayed.

The fields in the View/Modify User Models screen are the same as those on the Add Local Voice User - Change Defaults screen (see page 6-20) and the Outcalling Fields screen for local voice users (see page 6-46). See those pages for details. However, the first field is unique to this screen:

- ***Model Name*** - The name of the user model. The default is taken from the names displayed in the User Models list. If this field is blank, the model does not appear in the Add Local Voice User screen (Figure 6-4) and you will not be able to select it when adding users to the system.

Procedure 6-22xxx
Viewing/Modifying User Models

Starting point : View/Modify User Models screen.

1 Move the cursor to the field you wish to change.

2 Enter the data or select the appropriate field.

3 Press [Outcalling Fields] to modify the Outcalling parameters for user models. (See Figure 6-24.)

4 Select [Return to Basic Fields] to return to the View/Modify User Models screen.

5 Choose step 5a to save your changes or 5b to cancel.

a. Use [Save].

The user model is saved; the User Models List screen is redisplayed.

b. Use [Cancel].

Changes are discarded and the User Models List screen is redisplayed.

Figure 6-24xxx
View/Modify User Models screen (Outcalling Fields)

```
                          User Administration

 View/Modify User Models - Outcalling Fields

* Delivery to Non-Users Capability:  No[Yes]

* DNU DTMF confirmation required:    [Disabled] Enabled

* Delivery to non-user               None  On_Switch [Local] Long_distance_1
 restriction/permission codes       Long_distance_2

                        ------------------------------

 Remote Notification Capability:    [No] Yes

 Current State:                     Off

* Keypad Interface:                  Disabled  [Enabled]

 Message Remote Notification Option:[Any]  Urgent

 Remote Notification                None  On_Switch [Local] Long_distance_1
 restriction/permission codes       Long_distance_2

 Busy    Retry Limit:3__      Retry Interval (hh:mm):  00:05
 No Answer Retry Limit:10_   Retry Interval (hh:mm):  00:15
 Answered  Retry Limit:1__   Retry Interval (hh:mm):  00:05

 Business Days:      Sunday      [No] Yes
                     Monday       No  [Yes]
                     Tuesday      No  [Yes]
                     Wednesday    No  [Yes]
                     Thursday     No  [Yes]
                     Friday       No  [Yes]
                     Saturday    [No] Yes

 The Outcalling Fields data will be saved only if the model is saved.

  Return to
 Basic Fields
```

*   These fields are displayed only if Voice Messaging (MMUI) is enabled.

Distribution Lists

> ***Note:*** Distribution lists cannot be created if Call Answering (VMUIF) is installed on the system.

Distribution lists allow you to address the same voice message to more than one person at a time. (Users can also create personal distribution lists which serve the same purpose.) When the message is sent, it is deposited in every mailbox included in the list. Distribution lists are created and modified in the Distribution Lists screen (Figure 6-25).

You can create any number of distribution lists containing up to 120 entries each. You may find it easier to assign numbers to distribution lists that are of a different series from those used as mailbox numbers to avoid confusion or conflict. Ensure that distribution list numbers do not conflict with any dialing plan prefixes or codes. Users can create up to 9 personal distribution lists containing 99 entries each using their telephone keypad.

You can also record a list title for each distribution list that you create. The idea of a list title is similar to that of the personal verification. It is played when a distribution list number is entered when addressing messages. It is recommended that you record a list title, describing who is included in the list or the purpose of the list. This will make it easier to identify whether or not you have entered the correct list number when addressing messages.

Distribution lists can include the following types of numbers:

- mailbox numbers of local voice users
- mailbox numbers of remote voice users

Mailbox numbers at AMIS sites or remote sites in a Meridian network cannot be included in system distribution lists (if not defined as remote voice users). However private network addresses can be added to users' personal distribution lists. AMIS addresses can be added to personal distribution lists if the AMIS site is defined as a virtual node in your Meridian network. (See the chapter "AMIS Networking" for more information about virtual nodes.)

The following types of numbers do not have mailboxes associated with them and therefore can not be included in a distribution list:

- numbers of directory entry users
- remote notification targets
- delivery to non-user targets

Figure 6-25xxx
Distribution Lists screen

```
                         Distribution Lists

                 1 Add New Distribution List

                 2 Modify Distribution List

                 3 Delete Distribution List

                 4 View Distribution List titles

                 5 Print Distribution List titles

                 6 Print all Distribution List entries

Select an item >
  Exit
```

Procedure 6-23xxx
Navigating the Distribution Lists Menu

Starting point : User Administration Screen, <6> entered.

1 The ystem Distribution Lists screen appears (Figure 6-25).

2 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears. See the following sections for details:

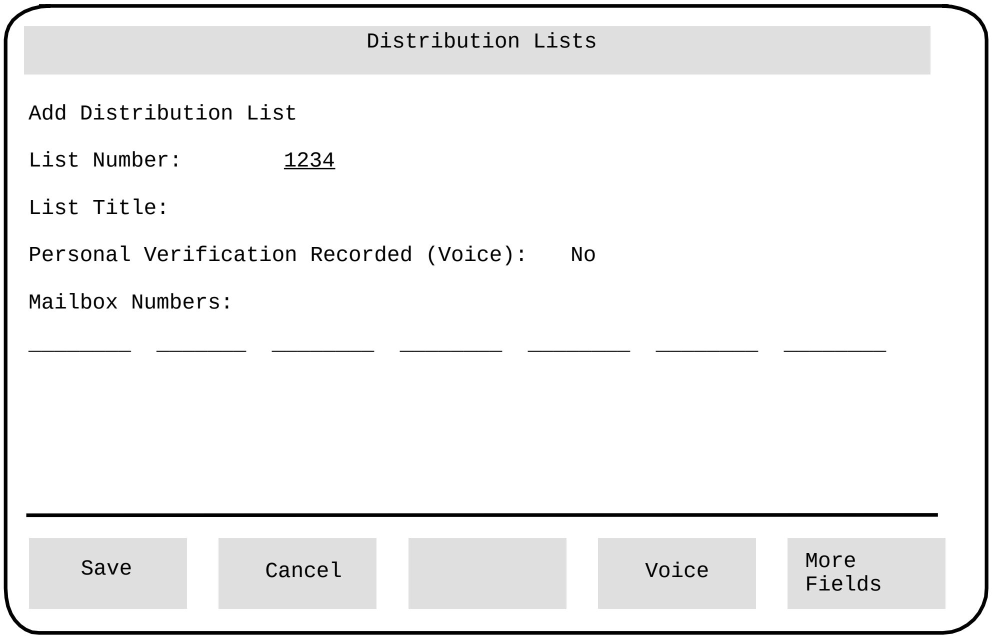
<1> "Creating distribution lists"
<2> "Modifying a Distribution List"
<3> "Deleting a Distribution List"
<4> "Viewing distribution list titles"
<5> "Printing list titles and entries"

3 Use [Exit] to return to the User Administration screen.

Creating distribution lists

Distribution lists are created at the Add Distribution List screen (Figure 6-26).

Figure 6-26xxx
The Add Distribution List screen



The screen shows the following fields:

- ***List Number*** - This value uniquely identifies the distribution list. The valid range is from 11 to 99999999 (only digits are allowed). The numbers 1 to 9 are reserved for user's personal distribution lists and cannot be used by administrators. The list number cannot be the same as the following numbers:
  - any mailbox number
  - the name dialing prefix (the default prefix is 11; do not use 11 to number a list unless you are sure that the name dialing prefix has been changed)
  - another distribution list number
  - any dialing plan access code prefixes
- ***List Title*** - The title of the distribution list, up to 41 characters in length. Do not use the special characters "+", "?", or "_". This field is blank by default. This title can also be used with Name Addressing when you compose and send a message.

- ***Personal Verification Recorded (Voice)*** - This field indicates that a spoken name has been recorded for this list. It is a good idea to record a personal verification for each distribution list. This will help you to identify the list after you have entered its number when composing a message. Choose a name that uniquely identifies this list. This field can be set only by using the [Voice] softkey. The default is “No”.
- ***Mailbox Numbers*** - Enter the mailbox numbers of the local voice users who are to be included in the distribution list. Each field holds up to 8 digits. Up to 120 mailbox numbers are allowed in a distribution list. By default, these fields are blank. The [More Fields] softkey can be used to add fields as additional mailboxes are required.

Procedure 6-24xxx
Adding a Distribution List

Starting point : Distribution Lists screen, <1> entered.

1 You are prompted to enter a number for the new list.

2 Enter a number that conforms to the rules described under the *List Number* field.

The Add Distribution List screen appears (see Figure 6-26) with the cursor positioned in the List Title field.

3 Enter a name for the list in *List Title*.

4 Enter the mailbox numbers of the users you want to include in the distribution list.

If you are entering the mailbox number of a remote voice user, enter the network prefix followed by the mailbox number.

5 Use [More Fields] if you have reached the last available *Mailbox Number* field and wish to add more mailboxes to the list. Up to 120 mailboxes can be included in a list.

6 Use [Voice] if you want to record a personal verification for this distribution list. The procedure for doing this is described on page 6-56.

7 Choose step 7a to save the distribution list or 7b to cancel.

a. Use [Save].

The distribution list is saved; if you have created a long distribution list, it may take a few moments to save.

You are prompted to enter a number for a new distribution list; proceed to step 2 if you wish to continue, or 7b to leave the screen.

b. Use [Cancel].

You are returned to the Distribution Lists screen.

Modifying a Distribution List

The following procedure describes how to modify an existing distribution list. The fields in the View/Modify Distribution List screen are identical to the fields described in “Creating distribution lists” on page 6-96, except that the *List Number* field is read-only.

Figure 6-27xxx
View/Modify Distribution Lists screen

```
                         Distribution Lists

View/Modify Distribution List

List Number:        6674

List Title:         Accounting

Personal Verification Recorded (Voice):   No

Mailbox Numbers:

 1234      56789     987654      13578     ________   ________
________

 Save       Cancel                 Voice      More
                                              Fields
```

Note: If you log on while another administrator is modifying the same distribution list, only the [Exit] softkey will be displayed.

Procedure 6-25xxx
Modifying a Distribution List

Starting point : Distribution Lists screen.

1 You are prompted to enter the distribution list number.

2 Enter the list number to be modified, then press <Return>.

The View/Modify Distribution List screen appears (Figure 6-27) with the cursor positioned in List Title.

3 Modify the list title if you wish.

4 Change, add, or delete any mailbox numbers by using the keyboard cursor keys.

To delete an entire list, see the next section, “Deleting a Distribution List”.

5 Use [More Fields] if you have reached the last available mailbox number and wish to add more mailboxes to the list. Up to 120 mailboxes can be included in a list.

6 Use [Voice] if you want to record a personal verification for this distribution list. The procedure for doing this is described on page 6-56.

7 Choose step 7a to save the distribution list or 7b to cancel.

a. Use [Save].

The distribution list is saved; if you have modified a long distribution list, it may take a few moments to save.

You are prompted to enter a number for another distribution list; proceed to step 2 if to continue, or 7b to leave the screen.

b. Use [Cancel].

You are returned to the Distribution Lists screen.

Deleting a Distribution List

The following procedure describes how you can delete an entire distribution list. Only the list is deleted; the mailboxes which comprise the list are not deleted. For field descriptions, see “Creating distribution lists” on page 6-96; all fields are read-only.

Figure 6-28xxx
Delete Distribution List screen

```
                         Distribution Lists

Delete Distribution List

List Number:        1001

List Title:         Accounting

Personal Verification Recorded (Voice):    No

Mailbox Numbers:

4455   4652   4239    4807

____________________________________________________________________

Select a softkey >

  OK to         Cancel
  delete
```

Note: If you log on while another administrator is modifying the same distribution list, only the [Exit] softkey will be displayed.

Procedure 6-26xxx
Deleting Distribution Lists

Starting point : System Distribution Lists screen, <3> entered.

1 You are prompted to enter the distribution list number.

2 Enter the number of the distribution list you want to delete followed by <Return>.

The Delete Distribution List screen appears (see Figure 6-28) with the cursor positioned in List Title.

3 Choose step 3a to delete the distribution list or 3b to cancel.

a. Use [OK to Delete].

The distribution list is deleted. If you delete a long distribution list, the operation may take a few moments to complete.

You are prompted to enter a number for another distribution list to delete; go to step 2 if you wish to continue or 3b to exit the screen.

b. Use [Cancel].

You are returned to the Distribution Lists screen.

Viewing distribution list titles

This function allows you to view an alphabetical listing of distribution list titles. Use the list to obtain the number of an existing distribution list, if you need to modify, delete, or print it.

Procedure 6-27xxx
Viewing Distribution List Titles

Starting point : System Distribution Lists screen, <4> entered.

1 A list of the distribution list numbers and titles appears.

2 Use [Next Page] to view the next page of the list.

When [Next Page] no longer appears, the end of the list has been reached.

3 Use [Exit] (at any time) to return to the Distribution Lists screen.

Printing list titles and entries

This procedure describes how to print a list of all the distribution list titles or entries on the system.

Procedure 6-28xxx
Printing distribution lists and entries

Starting point : Distribution Lists screen, <5> entered for titles or Distribution Lists screen, <6> entered for entries.

1 The following softkeys appear: [Continue Printing] and [Cancel Printing].

You are prompted to check that the printer is ready and on-line.

2 Choose step 2a to print the distribution list titles or 2b to cancel.

a. Use [Continue Printing].

The list of distribution list titles begins printing.

Once printing is complete, the Distribution Lists screen and its softkeys are redisplayed; you may stop printing at any time by proceeding to 2b.

b. Use [Cancel Printing].

The print operation is canceled, and you are returned to the Distribution Lists screen.

There may be some delay before control is returned to the screen because the system waits for the printer to stop.

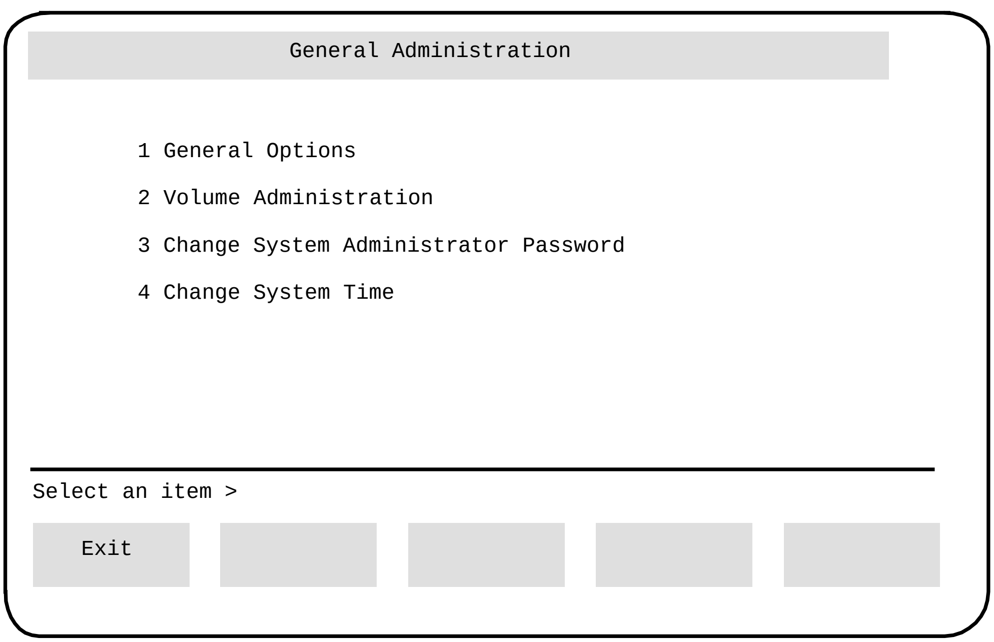
General Administration

The functions in the General Administration menu allow you to perform routine administrative tasks, such as backups, system time changes, and administrator password changes. From this menu you can also access the General Options screen from which you can define broad system characteristics such as printer port names, the date format, and the attendant DN.

The General Administration menu

The General Administration menu displays the options shown in Figure 7-1.

Figure 7-1
The General Administration menu



Procedure 7-1xxx
Navigating the General Administration menu

Starting point : Main Menu, <2> entered.

1 The General Administration screen appears (Figure 7-1).

2 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears.
See the following sections for details:

<1> “General Options”;
<2> “Volume Administration”;
<3> “Changing the system administrator password”;
<4> “Changing the system time”

3 Use [Exit] to return to the Main Menu.

General Options

The General Options screen contains parameters for configuring broad characteristics of your system.

Your DMS Voice Mail system will be equipped with one of the following features: Voice Messaging or (Simplified) Call Answering.

The feature that is installed determines:

- the user community to which users are added (private or residential),
- the interface that is presented to users (MMUI: the full-featured voice messaging interface or VMUIF: the simplified call answering interface),
- the features that are available to users, and
- the default user models.

See the chapter "Understanding DMS VoiceMail administration" for more information about the Voice Messaging (MMUI) and Call Answering (VMUIF) features.

Figure 7-2xxx
General Options screen

```
                    General Administration
General Options

Serial Number:          123456789

System Name:            DMS VoiceMail Customer

System Number:          0

Available Features:     Multi-Customer
                        Multiple Administration Terminals
                        SMDI
                        Meridian Access *
                        AdminPlus *
                        Voice Messaging
                        Call Answering
                        Dual Language Prompting
                        Voice Menus & Announcements
                        Outcalling
                        Voice Forms
                        Meridian Networking *
                        AMIS Networking

                                                  MORE BELOW
 Save      Cancel
```

* These are optional features that are available for CPE systems only.

Figure continued on next page.

Figure 7-2xxx
General Options screen (continued)

```
                    General Administration          MORE ABOVE

Attendant DN                 0__________________

Date Format for Administration
and Maintenance Reports:    mm/dd/yy [yy/mm/dd]   dd/mm/yy

SEER printing               Disabled  [Enabled]

SEER Printer Port Name:     ____________(blank implies console port)

Reports Printer Port Name: ____________(blank implies console port)

_____________________________________________________________________

    Save        Cancel
```

The following fields are displayed:

- ***Serial Number*** - This is a ten-digit read-only number which is used when placing software orders.
- ***System Name*** - This is the name by which DMS V oiceMail is identified to the switch. You may enter a name up to 30 alphanumeric characters in length. This field defaults to the name supplied during installation.
- ***System Number*** - Not applicable.
- ***Available Features*** **-** This list displays all of the features that are installed on the system. Figure 7-2 displays a list of all possible features for illustration purposes.

  The following features are optional and may or may not be installed. These fields are read-only.

  - Multi-Customer

    During DMS VoiceMail installation, either CO (central office) or CPE (central premise equipment) must be specified. If CO is selected, Multi-Customer is automatically enabled on the system. If CPE is selected, multi-customer is not enabled during software installation. (A feature expansion is required to install Multi-Customer on a CPE system.)

  - Multiple Administration Terminals

- SMDI (this is the Multi-SMDI feature which provides additional connectivity capability)
- Meridian Access/Unix Access (only available for CPE systems)
- AdminPlus (only available for CPE systems)

One of the following will be installed on your system:

- V oice Messaging
- Call Answering

The following features are optional and may or may not be installed:

- Dual Language Prompting

  Note: To see the languages installed on your system, see the Voice Messaging Options screen (described in the "Voice Administration" chapter).

- V oice Menus & Announcements
- Outcalling
- V oice Forms
- Meridian Networking (for CPE systems only).
- AMIS Networking

- ***Attendant DN*** - This field is not applicable if Call Answering (VMUIF) is installed on the system. This field indicates the extension number to which a caller is transferred when a user-customized revert to the operator is unsuccessful. (The custom revert is defined in the Add or Modify Local Voice User - Change Defaults screen. This is described in the "User Administration" chapter.) The number can be up to 30 digits and may begin with the digit "0". This field may be left blank. The default is "0".
- ***Date Format for Administration and Maintenance Reports*** - The format selected is used on reports generated by the MMI, including lists of users, operational measurement reports, and SEERs. It also specifies the format used for inputting dates. The default is yy/mm/dd. (Other possibilities are mm/dd/yy and dd/mm/yy.
- ***SEER Printing*** - When this field is "Enabled", System Error and Event Reports (SEERs) are printed as events or errors occur. If you do not have a printer, disable this feature. When this field is "Disabled" SEERs can only be viewed on screen. More detail is given when SEERs are printed than when they are displayed on screen. The default is "Enabled".

  Even when the system is working well and few error reports are generated, many event reports are produced. This means that the SEER buffer will fill up relatively quickly. Once full, contents are automatically deleted. It is therefore recommended that you print your SEERs on a regular basis. This will also help you troubleshoot problems as you will be able to look back through system events to monitor the beginning and history of a problem. If you are going to view SEERs on screen only, do so on a daily basis as critical information can be lost within a few days.

Note: You can also generate customized SEER reports by filling in the System Event and Error Reports screen in System Status and Maintenance. From this screen, you can view or print SEERs according to SEER class, SEER type (error, admin and system) or severity level (critical, major and minor). (If SEER printing is disabled in this field SEERs are still collected on disk and can be viewed.) See page 10-32 in the “System Status and Maintenance” chapter for more information.

- ***SEER Printer Port Name*** - The printer port to which the dedicated SEER printer is connected (if installed). This requires a data port on the MSP which must be defined as a printer port in the hardware database. This field holds up to 12 alphanumeric characters. If this field is left blank, reports will print to the console printer port.
- ***Reports Printer Port Name*** **-** This field indicates the printer port to which the dedicated printer for Operational Measurement reports, and general printing from the System Administration menus, is connected (if installed). This requires a data port on the MSP which must be defined as a printer port in the hardware database.) If this field is left blank, reports will print to the console printer port.

Procedure 7-2xxx
Modifying General Options

Starting point : General Administration screen, <1> entered.

1 The General Options screen appears (see Figure 7-2) with the cursor positioned in System Name.

2 Use the cursor keys to move the cursor to the field you wish to modify; make the required changes.

3 Choose step 3a to save the changes, or 3b to cancel.

a. Use [Save].

The changes are saved and the General Administration screen is displayed.

Note: *If you modify the system number, reboot the system for the change to take effect.*

b. Use [Cancel].

You are returned to the General Administration screen.

Volume Administration (tape backups)

Volumes are subdivisions of the overall storage capacity of a hard disk. DMS VoiceMail Volume Administration provides the capability to make backup copies of some or all of the data stored on a hard disk. If a disk fails, data can be restored from the backup so that the system can be brought back into service quickly with minimum loss of information.

A Field Support representative can restore a system to the state it was in at the time of the last backup. To ensure that this recovery process is complete, you should make certain that you have on hand a complete set of backup tapes. If no backups have been kept, a complete re-entry of all user and site-specific information will be required. How often you back up your data is influenced by how often changes are made to user and system information. If you make important changes to the system daily, then daily backups may be in order.

CAUTIONS
Backing up

Perform backups regularly. Recovery from a system where no backups have been kept implies a complete re-entry of all user and site-specific information.

Avoid backing up the system between the hours of 2:30 a.m. and 5:00 a.m. since important system audits take place during these hours.

Do not perform backups when the system is peaking above 50% of the rated capacity.

Store tapes in a secure area free of electromagnetic fields; store important backup tapes off-site for added security.

Do not use Northern Telecom software distribution tapes for backing up your system; these tapes are important in recovering from disk failures.

Do not re-use the same tapes for consecutive backups. It is recommended that you maintain at least two sets of backup tapes and that you use these sets in rotation.

Store tapes in their cases, label them clearly and set the write protection tab (turn the rotating knob until the arrow points to safe).

Full and partial backups

There are two types of backups that can be performed, partial and full.

- ***Partial Backup*** - When you perform a partial backup you save the administrative configuration of the system, including the user database, call answering greetings and spoken names, but not including the users' voice data (voice messages and greetings). Restoring from a partial backup avoids the need to re-enter all users. However, the voice messages and user greetings will be lost. For a partial backup the following volumes are backed up: VS1T, VS1V, VS1B and VS901T. See the next section for a description of the various volumes.
- ***Full Backup*** - All the information is copied, the system text and voice data and the user text and voice data (messages and greetings). Normally, full backups are not done because user messages and greetings are transitory and do not warrant the extra time required to back them up. However, if the loss of messages carries financial or legal implications, weekly or even daily backups of voice data may be warranted. For a full backup the following volumes are backed up: VS1T, VS1V, VS1B, and all the VSxT, VSxV, and VSxB volumes, where x is 203 through 210 depending on the number of nodes and type of disk packs.

  Note: VS1B is a temporary volume that is created during backup and is copied to tape.

Volume numbers and distribution

DMS VoiceMail systems can have from two to sixteen nodes. These are divided into: nodes 1 and 2, the MSPs (Multi-Server Processors), nodes 3 through 10, the SPNs (Signal Processing Nodes), nodes 11 and 12, reserved for future use, and nodes 13 -16, the TIFNs (Telephony Interface Nodes). See Table 7-1. Nodes 1 through 10 contain the hard disk drives for data storage with the disk drives being partitioned into volumes. Volumes are storage areas for system and user related information. The volumes are already set up when your system is installed. Table 7-2 specifies the maximum amount of storage available on each volume for the various DMS VoiceMail configurations.

When initially setting up DMS VoiceMail, you must distribute DMS VoiceMail users over the volumes by assigning a volume number to each user - see "Distributing users over volumes" and "Adding local voice users" in the "User Administration" chapter.

By convention, the system volume on the first node is named VS1, and the user volumes on the SPN nodes are named VS2xx, where xx is the number of the node on which the volume is located. Volumes are given numbers of the type “VStnnX”. The first digit in the volume number, t, indicates the type of information stored on the volume where:

1 system information
2 user information
9 partial backup user profiles

The last two digits in the volume number nn indicate the node number.

Each volume contains two “regions”. X will either be T (for text data) or V (for voice data).

For example, VS205T refers to the text region of a user volume on node 5.

The system volume, VS1, contains the following user information:

- each user’s personal verification
- organization profile
- customer profiles
- corporate directory
- operation measurement traffic and billing data
- program software
- network database
- voice menus and announcements (if installed)
- voice form definitions (if installed)
- network message queues
- voice prompts

The user volumes (VS203, VS204 ...VS210) may contain the following information:

- messages
- greetings
- voice menus and announcements and form definitions if these require more space than is available on VS1
- user information
- voice form responses

Voice menus, voice forms and announcements are located on VS1 or VS203. Check the Voice Services Profile screen to determine which volume contains menus and announcements. (See “The Voice Services Profile” in the “Voice Administration” chapter.) If these voice services are stored on VS203, you should do a Voice & Data backup of this volume on a regular basis. If you backup Data only, any greetings that you have recorded will not be backed up.

Information on disk usage can be obtained through the Disk Usage report (see “Traffic Reports” in the “Operational Measurements” chapter). Listings of the volumes is obtained by displaying the Volume Administration screen, described later in this chapter.

Table 7-1xxx
Volume distribution on the DMS VoiceMail nodes

Node 1	Node 3	Node 5	Node 7	Node 9
VS1T VS1V VS2T VS2V VS901T	VS203T VS203V	VS205T VS205V	VS207T VS207V	VS209T VS209V
Node 2	**Node 4**	**Node 6**	**Node 8**	**Node 10**
	VS204T VS204V	VS206T VS206V	VS208T VS208V	VS210T VS210V

TIFN Nodes are nodes 13, 14, 15 and 16 and do not have any volumes
VS1 contains voice prompt sets 3 and 4, and VS2 contains voice prompt sets 1 and 2.
T indicates text data.
V indicates voice data.

Table 7-2xxx
User disk capacities for DMS VoiceMail systems

SPN nodes		Maximum Hours available for Voice Storage (per disk volume)							
Size	**Hours**	VS203	VS204	VS205	VS206	VS207	VS208	VS209	VS210
2-node	**150** **300**	150 150	- 150						
4-node	**300** **600**	150 150	- 150	150 150	- 150				
6-node	**450** **900**	150 150	- 150	150 150	- 150	150 150	- 150		
8-node	**600** **1200**	150 150	- 150	150 150	- 150	150 150	- 150	150 150	- 150

Volumes recommended for regular backup

Back up the following volumes on a regular basis (usually weekly).

- VS1 [Voice & Data]
- VS203 ... VS210 [Data]

  Note: If voice menu applications (and/or voice forms) are stored on VS203, then do a [Voice & Data] backup of VS203.

To do a full backup of the entire system, backup the following volumes:

- VS1 [Voice & Data]
- VS203 ... VS210 [Voice & Data]

Scheduling a backup

Backups should be carried out at a time when the system is quiet or outside the regular hours for your organization. Do not backup the system when the system is operating above 50% of the rated capacity. Do not back up the system between the hours of 2:30 a.m. and 5:00 a.m. since important system audits take place during these hours. These audits are activated automatically at the same time every day and ensure continued operation of your system. Do not schedule a backup if more than one tape is required.

Backup tapes

All DMS VoiceMail systems have a tape drive capable of reading and writing industry standard 1/4-inch data cartridges. Both partial and full backups can be made to tape on all DMS VoiceMail systems.

DMS VoiceMail systems use Viper 2150S cartridge drives. The type of backup tape used is the DC6250 which can store 250 MB of data. (Although this tape drive accepts DC6150 (150 MB) tapes, it is not recommended that they be used.)

The approximate number of tapes required for one full and one partial backup are listed in Table 7-3.

Table 7-3xxx
Backup tape requirements*

Configuration	1 Full Backup	1 Partial Backup
2 SPN nodes	12/18 tapes	6 tapes
4 SPN nodes	18/30	6
6 SPN nodes	30/42	6
8 SPN nodes	30/54	6

* Represent approximate requirements.

Using the tape drive

DMS VoiceMail uses streaming tape drives which record data on multiple tracks on the tape. Each track runs from one end of the tape to the other. At the end of the tape, the tape head is positioned to the next track and the tape direction is reversed. After each block of data is written, it is read back and checked. If it cannot be correctly read, the data will be rewritten in the next block. After 16 unsuccessful attempts to write the data, a parity error is signaled and the backup fails. Such failures can be caused either by flaws in the tape or dirty tape heads. For information on cleaning the tape drive, see *Routine Maintenance Procedures* (NTP 297-7001-501).

Tape cartridges can be write-protected by turning the rotating knob until the arrow points to the Safe indicator. Any attempt to write on a write-protected cartridge will generate an error.

Performing backups: the Volume Administration screen

Data storage on the hard disk is distributed between volumes. Volumes are subdivisions of the hard disk. The Volume Administration screen (Figure 7-3) is used to back up the volumes. It displays all the volumes on your system, their designated use, their capacity in kilobytes and equivalent hours and minutes, and the percentage of voice and data storage currently used. To back up a volume, you select it and use the applicable softkey as described below.

Figure 7-3xxx
Volume Administration screen

```
                         General Administration


Volume Administration

Volume       Use              Volume Size                 Usage (% Full)
 Name                  Data          Voice                 Data    Voice
                     (KBytes)     (KBytes)  (hh:mm)

VS1          System    129992     14976     1:51           22       3
VS203        Users     30         200       2:00           13      23
VS204        Users     20         150       1:59           17      40
VS205        Users     30         200       2:00           14      17
...          ...       ...        ...       ...            ...     ...
VS210        Users     30         200       2:00           14      17

____________________________________________________________________________
Move the cursor to the desired volumes and press the space bar to select.

    Exit       Backup to                 Backup Status   View/Delete
                 Tape                                     Schedule
```

Procedure 7-3xxx
Performing a backup

Starting point : General Administration screen, <2> entered.

1 The Volume Administration screen appears (see Figure 7-3).

2 Use the cursor keys to move the cursor to the volume name you wish to back up; use <Space Bar> to select the volume. Repeat this for each required volume.

3 Choose step 3a to start a backup, 3b to monitor the progress of a backup, 3c to display the backup schedule, or 3d to return.

a. Use [Disk to Tape Backup].

See the section “Disk to Tape Backup”.

b. Use [Backup Status].

See the section “Backup Status” later in this chapter for details.

c. Use [View/Delete Backup Schedule].

See the section “View/Delete Backup Schedule” for details.

You can only change the backup schedule while in the Disk to Tape Backup function.

d. Use [Exit] to return to the General Administration menu.

You need not wait for a backup to complete before returning to other menus. The backup will proceed while you perform other tasks, and notify you if the backup process requires your attention.

Disk to Tape Backup

When a volume is selected and the [Backup to Tape] softkey on the volume administration screen is pressed, the Disk to Tape Backup screen (Figure 7-4) is displayed. The screen displays the volumes selected in the Volume Administration screen, and provides a set of backup options for each volume. For a partial backup, use the “Data” option on all of the user volumes. For a full back up, use the “Voice and Data” option on all of the user volumes. The system volume only has “Voice and Data” as a backup option.

The Disk to Tape Backup screen also allows you to automate backups through the [Schedule Backup] softkey. To use this function, insert a tape in the tape drive before the scheduled backup time; the backup will proceed automatically at the specified hour. You will be informed of how much data is to be copied when you schedule the backup.

> ***Note:*** Do not perform a scheduled backup to tape if more than one tape is required.

If a tape error occurs during backup, you do not have to restart the backup process from tape 1. Follow the instructions as they appear on the screen. In some instances you are required to keep the tape, as the data that was recorded is not corrupt; in other instances you will be required to discard the tape. At this stage you should clean the tape heads (as described in the *Routine Maintenance Procedures* (NTP 297-7001-501) before inserting another tape cartridge.

Figure 7-4xxx
Disk to Tape Backup screen

```
                         General Administration

Disk to Tape Backup


Volume
 Name                  Use               Backup Options

VS1                    System            Voice_&_Data
VS203                  Users             [Data] Voice_&_Data
VS204                  Users             [Data] Voice_&_Data
VS205                  Users             [Data] Voice_&_Data
...                    ...               ...    ...
VS210                  Users             [Data] Voice_&_Data

____________________________________________________________________

Select a softkey >

   Exit                        Schedule     Immediate
                               Backup       Backup
```

Procedure 7-4xxx
Performing a disk to tape backup

Starting point : Volume Administration screen, a volume selected and the [Backup to Tape] sofkey pressed.

1 The Disk to Tape Backup screen appears (Figure 7-4).

2 Ensure that the tape you are using is either blank or can be overwritten and that it is installed properly in the tape drive.

3 Use the cursor keys to move to the backup options of each volume and select the required options (Data or Voice & Data).

4 Choose step 4a to carry out a backup or 4b to schedule a backup for a later time.

a. Use [Immediate Backup].

The softkey display changes to [OK to Start Backup] and [Cancel].

You are prompted to insert a tape in the tape drive. You are told approximately how much data (in megabytes) will be backed up.

Use [OK to Start Backup] to initiate the backup or [Cancel] to return to the Disk to tape backup screen. Once a backup is started, the Backup Status screen appears; see "Backup Status" later in this chapter for details.

Before the backup proceeds, the tape is automatically retentioned.

If the tape is filled before the system is completely backed up, you are prompted to load another tape.

__Note__: A tape may still be rewinding even if the message on the screen indicates that the tape is completed. Do not remove the tape from the tape drive until it has finished rewinding.

When a tape is filled, the following message appears:

“Tape x (x is the tape number) completed. Insert new tape and press Continue Backup softkey”

__Note:__ If there is a tape error during backup, one of the following messages appears:

Keep tape and insert tape number n

where “n” is the number of the tape, or

Discard tape and insert tape number n

To continue the backup, remove the tape from the drive and insert a new tape. Press the [Continue Backup] softkey. Keep the tape that contains the error.

__Note:__ If you are working in another screen while a tape backup is in progress, the following message appears:

In progress backup requires new tape.

Go to the Backup status screen and use the softkeys as indicated below.

The following softkeys appear: [Continue Backup] and [Abort Backup]. To continue the backup, press [Continue Backup]; to cancel the backup, use [Abort Backup]; you are returned to the Volume Administration screen.

__Note:__ When a backup is completed, remove the tape and label it clearly; include the current date and time, tape number, and the volumes which were backed up.

Use [Abort Backup] to stop a backup from proceeding; you are returned to the Volume Administration screen.

b. Use [Schedule Backup].

See the section “Schedule Backup” for details.

5 Use [Exit] at any time to return to the Volume Administration screen.

Schedule Backup

The [Schedule Backup] softkey in the Disk-to-Tape backup screen displays the Schedule Backup screen (Figure 7-5). This screen allows you to set the backup frequency (daily, weekly, monthly) and start time. Once you have saved the backup schedule, backups proceed at the specified day and time. Return to this screen to make any necessary modifications to the existing backup schedule.

> ***Note:*** Do not schedule backups between 2:30 a.m. and 5:00 a.m. when important system audits occur. Do not schedule a backup if more than one tape is required; for this reason, only perform partial backup on a scheduled basis.

Figure 7-5xxx
Schedule Backup screen

```
                        General Administration

 Schedule Backup to Tape

 Backup Frequency:              Daily  [Weekly]  Monthly

* Weekly:                       [Sunday] Monday Tuesday Wednesday Thursday
                                Friday Saturday

**Day of Month:                 20

 Backup Start Time:             23:00

 Volumes selected for Backup:   VS1     Back up Voice & Data
                                VS203  Back up Voice & Data

 ____________________________________________________________________

   Save        Cancel
 Schedule
```

```
*   This line is displayed if Backup Frequency is set to Weekly.
**  This line is displayed if Backup Frequency is set to Monthly.
```

Procedure 7-5xxxx
Creating a Backup Schedule

Starting point : Disk to Tape Backup screen, [Schedule Backup] entered.

1 Move the cursor to the required backup frequency and press <Return>.

For weekly backups, the screen displays the days of the week; choose the day on which backups are to occur.

For monthly backups, the screen displays a prompt for the day on which backups are to occur; enter the required day.

2 Enter the backup start time.

3 Choose step 3a to save the schedule or 3b to cancel.

a. Use [Save Schedule].

The schedule is saved and you are returned to the Volume Administration screen; automatic backups are now in effect.

b. Use [Cancel].

You are returned to the Volume Administration screen.

Backup Status

The Backup Status screen (Figure 7-6) displays the current status of a backup, if one is in progress. The screen displays the start time, the volumes to be backed up, and the current progress on each volume.

Figure 7-6xxx
Backup Status screen

```
                         General Administration

Backup Status

Backup Started:          3/20/91   23:00

Backup Completed:

Backing up Volumes:      VS1     VS2     VS202

                         VS1T    80%          done
                         VS1V     0%          done
                         VS1B    80%          done *

------------------------------------------------------------------------

 Exit                                                           Abort
                                                                Backup
```

* VS1B is a temporary volume created during backup. Once the backup is complete, this volume is automatically deleted from the disk.

Procedure 7-6xxx
Displaying the status of the current backup

Starting point : Disk to Tape Backup screen, [OK to Start Backup] entered, or Volume Administration screen, [Backup Status] entered.

1 Choose step 1a to stop a backup in progress or 1b to return.

a. Use [Abort Backup]. This softkey is displayed only if a backup is in progress.

The backup is stopped and for backup to tape, the tape is rewound.

The Volume Administration screen is redisplayed.

See the section “Disk to Tape Backup”.

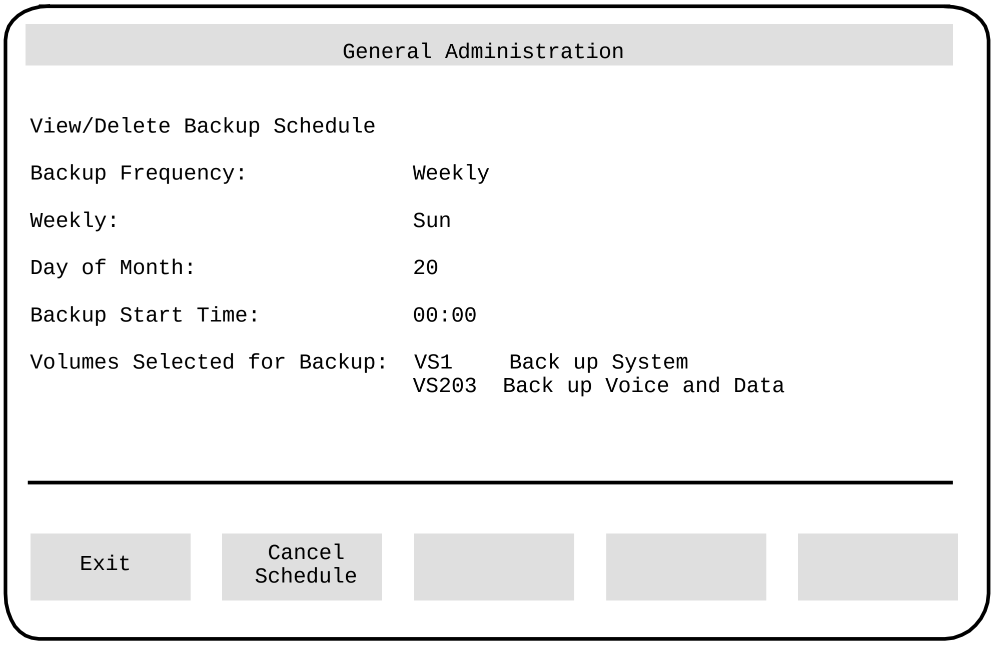
b. Use [Exit].

The Volume Administration screen is re-displayed; the backup continues.

View/Delete Backup Schedule

The View/Delete Backup Schedule screen (Figure 7-7) displays the current schedule for backups, if one exists. The screen is read-only and displays the current settings of the backup schedule, including the type of backup, how frequently backups are performed, the start time, the volumes to be backed up, and the backup options for each volume.

Figure 7-7xxx
View/Delete Backup Schedule



Procedure 7-7xxx
Viewing/Deleting Backup Schedules

Starting point : Volume Administration screen, [View/Delete Schedule] entered.

1 Choose step 1a to return to the Volume Administration screen or 1b to delete a schedule.

a. Use [Exit].

See the section “Disk to Tape Backup”.

The Volume Administration screen is redisplayed.

b. Use [Cancel Schedule].

The present schedule is deleted. No backups will occur at the scheduled time indicated. The Volume Administration screen is redisplayed.

Changing the system administrator password

When the system is first installed you are given a default system administrator password (**adminpwd**). When you log on for the first time, you are prompted for a new password. For security purposes, you should continue to change it on a regular basis. Passwords are not case-sensitive; any capitalization used in defining the password need not be used when entering the password. The maximum length is 16 digits. It is recommended that your administration password be at least 7 digits for added security.

Procedure 7-8xxx
Changing the administrator password

Starting point : General Administration Menu, <3> entered.

Note: The passwords are not displayed on the screen as you enter them.

1 You are prompted to enter the existing administrator password.

2 Enter the existing password.

3 You are prompted to enter the new password.

4 Enter the new password.

The system administrator password is alphanumeric (it can contain both letters and numbers) and must be between 1 and 16 digits in length.

5 You are prompted to enter the new password again, for verification purposes.

The new password is recorded and you are returned to the General Administration screen.

Changing the system time

The setting of the system clock in your DMS VoiceMail system should be accurate to keep correct records of events in your system (such as message creation and reception times or system event and error times).

Procedure 7-9xxx
Changing the system time

Starting point : General Administration Menu, <4> entered.

1 You are prompted to enter the new date and time.

2 Enter the date and time, followed by <Return>.

The clock is synchronized to the clocking signals from the network, the time is recorded, and you are then returned to the General Administration screen.

Voice Administration

DMS VoiceMail Voice Administration comprises all facilities related to processing voice information. These facilities, voice services, offer a range of functions from the simple playback of a recorded announcement to the more sophisticated voice menu applications, such as the automated attendant service. This section discusses voice services administration under the following headings:

- ***Voice Messaging Options*** - These parameters determine the general characteristics of the voice messaging service. If Voice Messaging (MMUI) is installed on the system, this includes configuring the broadcast mailbox number, the maximum delay for timed delivery, the name dialing prefix, the maximum message length and the maximum read message retention. The custom call answering greeting is also recorded in this screen (once for each language that is installed on the system). If Call Answering (VMUIF) is installed on the system, you will configure the lockout revert DN, the maximum message length and the maximum read message retention. From this screen, you will also record any introductory tutorials which describe the call answering service to new subscribers and the login greeting.
- ***Voice Security Options*** - These parameters allow you to control the level of security provided to users of DMS VoiceMail. For example, you can set the maximum number of invalid logon attempts that are allowed before a user's mailbox is disabled as well as several parameters related to user passwords. This is also where restriction/permission codes are defined. These codes are applied to various features and are intended to protect your system by preventing users and callers from placing calls (such as long distance calls) while connected to DMS Voice Mail.
- ***Voice System Configuration*** - allows you to configure voice services, either during initial installation or later, to accommodate additional services and voice channels.
- ***Voice Menu Applications Administration*** - allows you to create custom applications that offer a range of functions from the simple playback of a recorded announcement to the more sophisticated voice menus which allow callers to make choices by pressing keys on their telephone keypads and automated attendants which take calls during off-hours or holidays.

- ***Outcalling Administration*** - allows you to specify outcalling parameters which affect the remote notification and delivery to non-user features. You can also view the outcalling audit trail report to monitor the progress of remote notification and delivery to non-user calls.
- ***Voice Form Definitions*** - allow you to develop custom applications that ask specific questions of callers and collect their voice responses. These applications can be thought of as the electronic equivalent of the traditional paper form or questionnaire.

The topics covered under Voice System Configuration and Voice Menu Applications Administration are more complex than other topics covered in this document. The operations described under these headings should only be carried out by administrators trained for those specific tasks.

The Voice Administration menu

The Voice Administration menu (Figure 8-1) is displayed by selecting option <3> from the Main Menu.

> **CAUTION**
> **Overnight system audits**
> You should not leave the administrative console in any Voice Administration menu overnight or important system audits may fail due to a lack of available memory.

Figure 8-1xxx
The Voice Administration Menu

```
                         Voice Administration



     1  Voice Messaging Options

     2  Voice Security Options

   * 3  Voice System Configuration/Voice Menu Applications Administration

   **4  Outcalling Administration

   **5  Voice Form Definitions



 Select an item >
     Exit
```

```
* If the Voice Menu Application feature is not installed, this item will
  be "Voice System Configuration"
**Indicates optional features which may not be installed on your system.
```

Procedure 8-1xxx
Selecting items from the Voice Administration Menu

Starting point : Main Menu, <3> entered.

1 The Voice Administration menu appears (Figure 8-1).

2 Select an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears. See the following sections for details:

<1> "Voice Messaging Options";

<2> "Voice Security Options;

<3> "Voice System Configuration/Voice Menu Applications Administration"

<4> "Outcalling Administration";

<5> "Voice Form Definitions"

3 Use [Exit] to return to the Main Menu menu.

Voice Messaging Options

The Voice Messaging Options screen allows you to set voice messaging parameters. If Voice Messaging is installed, this includes setting the broadcast mailbox number, the maximum allowed delay for time delivery, the name dialing prefix and the maximum message length. The custom call answering greeting is also recorded in this screen. For multilingual systems, you can record a custom call answering greeting in all of the languages that are installed on your system. If Call Answering is installed, this means recording various greetings (tutorials and the login greeting), setting the maximum message length, the lockout revert DN and the maximum read message retention.

User interfaces

The first field in the Voice Messaging Options screen determines the telephone interface that is presented to users. It is a read-only field and the selection shown here depends on a setting in the General Options screen. When Voice Messaging is selected in the General Options screen, this field will be set to MMUI. When Call Answering is selected in the General Options screen, this field will be set to VMUIF.

The VMUIF Interface

This interface is intended for the Central Office that wishes to offer residential and small business subscribers a call answering and message retrieval service. It can also be used in a CPE environment to provide only basic call answering to users. Users of the VMUIF (Voice Messaging User Interface Forum) interface are known as *subscribers* and do not have access to voice messaging capabilities. The VMUIF interface is used by the (Simplified) Call Answering feature.

The MMUI Interface

The MMUI interface (Meridian Mail User Interface) is the Northern Telecom proprietary user interface. It provides users with:

- *call answering functionality* so that when a user is away from his or her phone, a caller can leave a message;
- *voice messaging functionality* which allows users to compose, send and manipulate voice messages;
- *advanced features* such as call sender, remote notification, delivery to non-users, name dialing and distribution lists.

The MMUI interface is used by the Voice Messaging feature. In a CPE environment, this is the more common interface. In a CO environment, this interface allows the service provider to meet the needs of Centrex customers or those subscribers that require full-featured voice messaging.

This screen will display different fields depending on a) which interface is selected (MMUI or VMUIF) and b) whether or not the multilingual system is installed. Figure 8-2 displays the screen for a non-multilingual system with the MMUI interface. Figure 8-3 displays the screen for a multilingual system with the MMUI interface. Figure 8-4 displays the screen for a non-multilingual system with the VMUIF interface and Figure 8-5 displays the screen for a multilingual system with the VMUIF interface.

Figure 8-2xxx
The Voice Messaging Options screen for non-multilingual systems (MMUI interface)

```
                     Voice Administration

 Voice Messaging Options

 Voice Messaging Interface:                        MMUI VMUIF

 Customized recording and Recording Selection for AmericanEnglish:

 Call Answering Greeting (Voice):                  No

 Broadcast Mailbox Number:                         12345678

 Broadcast Mailbox Personal Verification (Voice):No

 User Changeable Personal Verification:            Disabled [Enabled]

 Maximum Delay for Timed Delivery (days):          31

 Name Dialing and Name Addressing                  Disabled [Enabled]

 Prefix for Name Dialing and Name Addressing:      11

*Default Message Delivery Priority:                Standard [Economy]

 Maximum Message Length (mm:ss)                    03:00

 Mailbox Full Warning Threshold (percentage):      0

 Maximum Read Message Retention (days):            07
 ("0" implies that there is no system
 maximum limit. Read Message Retention will
 be determined from each user's profile.)

________________________________________________________________
Select a softkey >
   Save        Cancel                     Voice
```

* This field is displayed only if Meridian Networking is installed.

Figure 8-3xxx
The Voice Messaging Options screen for multilingual systems (MMUI interface)

```
                              Voice Administration
  Voice Messaging Options

  Default Language:                              [AmericanEnglish] Swedish

* Secondary Default Language:                    [AmericanEnglish] Swedish

  Default Language Overrides User's
  Preferred Language:                            [No] Yes

  Voice Messaging Interface:                     MMUI VMUIF

  Customized recording and Recording Selection for AmericanEnglish:

     Call Answering Greeting (Voice):            No

  Customized recording and Recording Selection for Swedish:

     Call Answering Greeting (Voice):            No

  Customized recording and Recording Selection for CanadianFrench:

     Call Answering Greeting (Voice):            Yes

  Customized recording and Recording Selection for Japanese:

     Call Answering Greeting (Voice):            No

  Broadcast Mailbox Number:                      12345678

  Broadcast Mailbox Personal Verification (Voice):No

  User Changeable Personal Verification:         Disabled [Enabled]

  Maximum Delay for Timed Delivery (days):       31

  Name Dialing and Name Addressing               Disabled [Enabled]

  Prefix for Name Dialing and Name Addressing:   11

**Default Message Delivery Priority:             Standard [Economy]

  Maximum Message Length (mm:ss)                 03:00

  Mailbox Full Warning Threshold (percentage):   0

  Maximum Read Message Retention (days):         31
  ("0" implies that there is no system
  maximum limit. Read Message Retention will
  be determined from each user's profile.)

  Select a softkey >
     Save        Cancel                    Voice
```

* This field is displayed only if Dual Language Prompting is installed.
**This field is displayed only if Meridian Networking is installed.

The following fields are displayed if the MMUI interface is enabled. The first three fields are displayed only on multilingual systems. (For VMUIF field descriptions, see page 8-13.)

- ***Default Language -*** This field lists all of the languages that are installed on the system. The primary default language (the first language installed) is highlighted. The selection made here determines the language in which prompts are played to callers during call answering and express messaging.
- ***Secondary Default Language -*** This field is available if the Dual Language Prompting feature is installed. The selection made here determines the secondary language in which prompts are played to callers during call answering and express messaging sessions. Prompts are played in the secondary language after being played in the language specified in the *Default Language* field.
- ***Default Language Overrides User's Preferred Language -*** When two or more languages are installed, users can specify a "preferred" language which is different from the default language. (The user's preferred language is defined in the Add Local Voice User screen.) However, if this field is set to "Yes", the language specified in the *Default Language* field overrides the user's preference.

  This field affects only those prompts that are played to callers during call answering and express messaging sessions. It does not apply to the prompts that users hear while logged on to DMS VoiceMail. These will still be played in the user's preferred language. For example, if the default language is AmericanEnglish, and a user's preferred language is Mandarin, the user will still hear DMS VoiceMail prompts in Mandarin; however, callers that reach the user's mailbox will hear prompts in English. When this field is set to "No", callers will hear DMS VoiceMail prompts in the preferred language of the user they have called.
- ***Voice Messaging Interface -*** This is a read-only field. MMUI is compatible with full-featured voice messaging.
- ***Customized Recordings and Recording Selections for <language> -*** On multilingual systems, this field is displayed once for each language that is installed.
- ***Call Answering Greeting (Voice) -*** This field indicates whether or not a custom call answering greeting has been recorded. The call answering greeting is played to external callers when they are connected to a user's mailbox through call answering or express messaging. This greeting is played before any personal greetings and typically contains the spoken name of the organization. To make a custom greeting, use the [Voice] softkey at the bottom of this screen. If you do not record your own greeting, no call answering greeting is played (there is no default greeting).

- ***Broadcast Mailbox Number*** - A broadcast message is a voice message that is delivered to all users in the system. In order to send a broadcast message, you (or a user) must specify a special mailbox number (the broadcast mailbox number) when composing the broadcast message. The default mailbox number is “999”.

  If this default number conflicts with the ESN access code and causes a conflict (e.g., if “9” is used as the ESN access code), change the broadcast mailbox number. If you change the broadcast mailbox number for this reason (or any other reason), ensure that the new number does not conflict with other numbers in the system.

- ***Broadcast Mailbox Personal Verification (Voice) -*** This field indicates whether or not a spoken name has been recorded for the broadcast mailbox number. This verification is announced to users before the message is played. It should inform users that the message they are about to hear is a broadcast message (and who it is from, if users may need to get in touch with the sender).
- ***User Changeable Personal Verification*** - When this field is enabled, users can record their own personal verifications through their telephone sets. When “Disabled”, only the administrator can record personal verifications. The default is “Disabled”.
- ***Maximum Delay for Timed Delivery*** - This field displays the maximum number of days that a message can be delayed before being delivered. The valid range is from 0 to 365 days. The default is “31”. If this field is set to “0” timed delivery of messages will not be available to users.
- ***Name Dialing and Name Addressing*** - This field allows you to disable the name dialing and name addressing features. These features should be disabled in those countries where the telephone keypads do not map to an alphabetical sequence recognizable to DMS Voice Mail. This field defaults to “Enabled”.

  Note: If you disable name dialing and name addressing and then re-enable them, the prefix for name dialing and name addressing is changed from the current value to null. Be sure to enter the correct prefix after re-enabling these features.

- ***Prefix for Name Dialing and Name Addressing*** - This field defines the prefix that users must dial in order to use name dialing or name addressing. The valid range is from 1 to 99. The (recommended) default is “11”.

  Note: Check that this number does not conflict with any of the following:

  - mailbox numbers (including the broadcast mailbox number)
  - telephone extensions

- distribution list numbers
- the DNU prefix
- location prefixes

These numbers conflict if their first two digits match the name dialing prefix.

Note: If name dialing and name addressing were disabled and then re-enabled, this field is reset to null.

- ***Default Message Delivery Priority -*** This field is displayed only when Meridian Networking is installed. Your choice determines when messages are delivered across a network. When “Standard” is selected, messages are retained for a certain period of time before they are sent to remote sites. If ”Economy” is specified, messages are sent at a specific time each day (usually off hours). The Standard holding time and Economy initiation time are set in the Network Scheduling Parameters screen, accessible through the Network Administration Menu. The default is “Standard”.
- ***Maximum Message Length (mm:ss)*** - This field determines the maximum message (or greeting) length that a user is allowed to record. You may enter a value from 00:30 to 99:00. The default is “03:00” (i.e., 3 minutes for each message or greeting).

  Note: For internal or external greetings, the maximum length allowed is 5 minutes (05:00), regardless of the value entered in this field. However, the maximum combined length allowed for both the internal and external greeting is also 5 minutes (not 10 minutes). If the internal greeting plus the external greeting together total more than 5 minutes, the last recorded greeting may not be saved.

- ***Mailbox Full Warning Threshold (percentage) -*** This field allows you to determine how full a user’s mailbox must become before the system plays the mailbox full prompt when the user logs on. A value of “0” means that the user will never hear the mailbox full warning prompt. The valid range is 0 to 100 (percent). The default is “0”.

  Note: A user may inform you that he or she has received the mailbox full warning, but that the mailbox is definitely not full. For example, the user is certain that there are only two short messages in the mailbox. A prematurely full mailbox is caused by an unexpected system reboot that leaves inconsistencies between the volume server and what is actually in the mailbox. This problem will be fixed automatically during the scheduled nightly audit. However, if an unexpected reboot happens at a busy traffic time, you can log on at the Tools level and select the menu item “Audit all volumes”. This will update the real mailbox storage information that is stored on disk and prevent prematurely full mailboxes. See the *System Administration Tools Guide* (NTP 555-7001-305) for more information about this tool.

- ***Maximum Read Message Retention (days)*** - This field determines the maximum number of days that messages will be kept in the user's mailbox after being read. When the maximum is reached, read messages are deleted. The valid range is from 0 to 31 days. If this field is set to "0", messages are not deleted by the system and are retained until deleted by the user. The default is "7" days.

  Note: The read message retention limit can also be configured for each user in the Add or Modify Local Voice User screen (see "User Administration"). The user's limit is overridden by the limit defined here (if a non-zero value).

Figure 8-4xxx
The Voice Messaging Options screen for non-multilingual systems (VMUIF interface)

```
                              Voice Administration
Voice Messaging Options

Voice Messaging Interface:                    MMUI VMUIF

Customized recordings and Recording Selections for AmericanEnglish:;

  VMUIF Introductory Tutorial (Voice): No    Type:None [Default]

  VMUIF Introductory Tutorial for
  Dial Pulse (Voice):                         Yes  Type:None Default [Custom]

  Login Greeting (Voice):                     No   Type:None [Default]

Lockout Revert DN:                            ________________________
(Blank implies no revert)

Maximum Message Length (mm:ss)                03:00

Maximum Read Message Retention (days):        31
("0" implies that there is no system
maximum limit. Read Message Retention will
be determined from each user's profile.)

   Select a softkey >

     Save       Cancel                   Voice
```

Figure 8-5xxx
The Voice Messaging Options screen for multilingual systems (VMUIF interface)

```
                      Voice Administration
Voice Messaging Options

Default Language:                      [AmericanEnglish] Swedish

Secondary Default Language:            [AmericanEnglish] Swedish

Default Language Overrides User's
Preferred Language:                       [No] Yes

Voice Messaging Interface:                MMUI VMUIF

Customized recordings and Recording Selections for AmericanEnglish:

VMUIF Introductory Tutorial (Voice): No   Type:None [Default]

VMUIF Introductory Tutorial for
Dial Pulse (Voice):                  No   Type:None [Default]

Login Greeting (Voice):              Yes  Type:None Default[Custom]

Customized recordings and Recording Selections for Swedish:

VMUIF Introductory Tutorial (Voice): No   Type:None [Default]

VMUIF Introductory Tutorial for
Dial Pulse (Voice):                  Yes  Type:None Default [Custom]

Login Greeting (Voice):              No   Type:None [Default]

Customized recordings and Recording Selections for CanadianFrench:

VMUIF Introductory Tutorial (Voice): No   Type:None [Default]

VMUIF Introductory Tutorial for
Dial Pulse (Voice):                  No   Type:None [Default]

Login Greeting (Voice):              No   Type:None [Default]

Lockout Revert DN:                   _____________________________
(Blank implies no revert)

Maximum Message Length (mm:ss)              03:00

Maximum Read Message Retention (days):      31
("0" implies that there is no system
maximum limit. Read Message Retention will
be determined from each user's profile.)

Select a softkey >
  Save       Cancel                  Voice
```

The following fields are displayed when the VMUIF interface is enabled. The first three fields appear only on multilingual systems.

- ***Default Language* -** This field lists all of the languages that have been installed on your system. The primary default language (the first language installed) is highlighted. The selection made here determines the language in which prompts are played to callers during call answering and express messaging sessions.
- ***Secondary Default Language* -** This field is available if the Dual Language Prompting feature is installed. The selection made here determines the secondary language in which prompts are heard by callers during call answering and express messaging sessions. Prompts are played in the secondary language after being played in the language specified in the *Default Language* field.
- ***Default Language Overrides User's Preferred Language* -** If this field is set to "Yes", the language specified in the *Default Language* field overrides the user's preference which is set in the Add Local Voice User screen. This means that all callers will hear the default language during call answering and express messaging sessions. However, it does not apply to the prompts that users hear while logged on to DMS VoiceMail. These will still be played in the user's preferred language. For example, if the default language is AmericanEnglish, and a user's preferred language is Mandarin, the user will still hear DMS VoiceMail prompts in Mandarin; however, callers that reach his or her mailbox will hear prompts in English. When this field is set to "No", callers will hear DMS VoiceMail prompts in the preferred language of the user they have called.
- ***Voice Messaging Interface* -** This is a read-only field. The interface (MMUI or VMUIF) determines the telephone set interface to which users have access. The interface is chosen when the system is ordered and can not be changed by the administrator. The Meridian Mail User Interface (MMUI) corresponds to the Voice Messaging feature and the Voice Messaging User Interface Forum (VMUIF) corresponds to the (Simplified) Call Answering feature.
- ***Voice Messaging Interface* -** This is a read-only field. VMUIF is compatible with simplified call answering.
- ***Customized recording and Recording Selection for <language>* -** These fields are displayed once for each language that is installed. The standard is American English.
- ***VMUIF Introductory Tutorial (Voice)* -** This field indicates whether or not a voice recording has been made for the introductory tutorial. The introductory tutorial is played to subscribers when they log on for the first time in order to familiarize them with the service. If a recording is made, the following field, *VMUIF Introductory Tutorial Type*, will allow you to select "Custom".

- ***VMUIF Introductory Tutorial Type* -** This field identifies the type of introductory tutorial to be played the first time a subscriber logs into a new mailbox. The “Custom” option is available if there is a voice recording of the introductory tutorial. If you do not record a custom tutorial, you can select the default recording. You also have the option of not playing an introductory tutorial at all.
- ***VMUIF Introductory Tutorial for Dial Pulse (Voice)* -** This field indicates whether or not a voice recording has been made for the tutorial for dial pulse users. If a recording is made, the following field, *VMUIF Introductory Tutorial for Dial Pulse Type*, will allow you to select “Custom”.
- ***VMUIF Introductory Tutorial for Dial Pulse Type* -** This field identifies the type of introductory tutorial to be played the first time a subscriber logs into a new mailbox from a dial pulse (rotary) telephone. The “Custom” option is available if you have recorded your own custom tutorial. If you have not recorded a custom tutorial, you can choose to play the default tutorial or no tutorial at all.
- ***Login Greeting (Voice)* -** This field indicates whether or not a voice recording has been made for the Login Greeting. This is the greeting that is played when subscribers log onto DMS Voice Mail.
- ***Login Greeting Type* -** This field determines which greeting is used, if there is one. If a custom login greeting has been recorded (see the previous field), you may select “Custom”. If one hasn’t been recorded, you can use the default greeting or select “None”.
- ***Lockout Revert DN* -** This field specifies the DN to which callers are reverted when the dialed mailbox is disabled (for example, after the subscriber has made too many invalid logon attempts). When this field is left blank, a prompt is played asking callers to try again at a later time.
- ***Maximum Message Length (mm:ss)*** - This field determines the maximum allowed length of messages left by callers in subscribers’ mailboxes. You may enter a value from 00:30 to 99:00 in 10 second increments. The default is “03:00”.
- ***Maximum Read Message Retention (days)*** - This field determines the maximum number of days that messages will be kept in the user’s mailbox after being read. When the maximum is reached, read messages are deleted. The valid range is from 0 to 31 days. If this field is set to “0”, messages are not deleted by the system and are retained until deleted by the user. The default is “7” days. For Dial Pulse residential subscribers, the default is “3” days.

  Note: The read message retention limit can also be configured for each subscriber in the Add or Modify Local Voice User screen (see the “User Administration” chapter). The user’s limit is overridden by the limit defined here (if a non-zero value).

Procedure 8-2xxx
Modifying Voice Messaging Parameters

Starting point : Voice Administration menu, <1> entered.

1 The Voice Messaging Options screen appears (Figure 8-2 through 8-5).

2 Move the cursor to the field you wish to modify and make the required changes.

3 Choose step 3a to save the changes or 3b to cancel.

a. Use [Save]. The changes are saved and you are returned to the Voice Administration menu.

b. Use [Cancel].

Changes are discarded. The Voice Administration menu reappears.

Voice Security Options

The Voice Security Options screen (Figure 8-6) allows you to control various security features and set restriction and permission codes that can be applied to features such as call answering, call sender, express messaging, mailbox thru-dial, AMIS networking, remote notification and delivery to non-users.

Restriction/permission codes

Restriction/permission codes are defined in the Voice Security Options screen. Up to four sets of codes can be created. Each set can be thought of as a restriction/permission table that defines which dialing codes are allowed and which are restricted. A dialing code can be up to 5 digits in length and can be one of the following: an access code (for dialing out of the switch, such as “9” for local calls and “91” for long distance calls), an area code, or a country code (area codes and country codes must be preceded by the appropriate access code, such as “91416” since “91” is needed to dial out of the switch). An internal extension can also be entered as a dialing code if you wish to restrict certain DNs (such as the President’s extension). Each table can contain up to 10 restriction codes and 10 permission codes.

Important: To ensure the security of your system, apply the appropriate restriction codes to the features listed for MMUI and VMUIF. This prohibits external callers or internal users from placing certain types of calls (such as local or long-distance calls). If local or long distance codes are not restricted and a caller or user places a call using one of the features described below, you will be charged for the call since the call will have originated from your switch.

For MMUI

Apply a restriction/permission table to the following features in the Add a Local Voice User screen.

- custom revert - to restrict the extension to which callers can be reverted when they press “0” while connected to DMS Voice Mail
- extension dialing - to restrict the extensions that users can dial
- external call sender - to restrict users from using call sender to dial certain numbers
- AMIS networking - to restrict outgoing AMIS messages
- remote notification (outcalling fields) - to restrict the target DNs to which remote notifications can be sent (such as long distance)
- delivery to non-users (outcalling fields) - to restrict the numbers to which messages to non-users can be sent (such as long-distance)

Apply a restriction/permission table to the following features in the Voice Security Options screen:

- call answering thru dial - to restrict the numbers that can be dialed by callers during call answering sessions if they try to thru-dial to another number while connected to DMS Voice Mail
- express messaging thru dial - to restrict the numbers that can be dialed by users during express messaging sessions

  Note: While someone is involved in a call answering or express messaging session, they can place a call by pressing “0” followed by an internal extension or external number. This is referred to as thru dial and should not be confused with thru-dialers which are one type of voice menu application.

- thru-dialers - For each thru-dialer that you create, select one of the restriction/permission tables that is defined in the Voice Security Options screen or create a custom restriction/permission table for the thru-dialer in the Add a Thru-Dialer Definition screen.

For VMUIF

Apply a restriction/permission table to the following features in the Add a Local Voice User screen.

- custom revert - to restrict the extension to which callers can be reverted when they press “0” while connected to DMS Voice Mail.
- remote notification (outcalling fields) - to restrict the target DNs to which remote notifications can be sent (such as long distance)
- thru-dialers - For each thru-dialer that you create, select one of the restriction/permission tables that is defined in the Voice Security Options screen or create a custom restriction/permission table for the thru-dialer in the Add a Thru-Dialer Definition screen.

Creating a restriction/permission table

You can create four separate restriction/permission tables in the Voice Security Options screen. However, for any feature (except thru-dialers) you can only apply one of the four tables that you define here. There are therefore different ways to approach restriction/permission codes. For example, you can create one table that only contains extension DNs that reside on the switch; a second table that restricts local calls; a third that restricts long distance calls and perhaps a fourth that restricts all local and long distance calls for instances where security is very important. It is, however, up to you to decide on the types of restriction/permission tables that you require.

Restriction/permission codes are entered in the field *List Names and Codes* in the Voice Security Options screen (see Figure 8-6).

CAUTION
All features are initially restricted
When DMS Voice Mail is installed, all 10 restriction fields are filled in. The first restriction code is defined as 0, the second is 1 and so on to the tenth code which is defined as 9. This means that all possible extensions and phone numbers are restricted and, therefore, all of the features to which you can apply restriction/permission codes will not work.

If you do not change the restriction/permission tables to permit certain numbers, the following features will not work:

- when callers press “0” they will not be reverted to the custom revert DN because “0” is restricted;
- users will not be able to dial any extensions;
- call sender will not work;
- users will not be able to send AMIS messages (although they will be able to receive them);
- users will not be remotely notified of their messages
- users will not be able to send messages to non-users
- callers will not be able to thrudial during call answering or express messaging sessions
- thru-dialers will not work

Defining restriction and permission codes

Example: You want to create a restriction/permission table that restricts all long-distance calls except calls to the area code 416 and calls to 911. The access code for making long distance calls is “91”. Fill out the restriction/permission table as shown below.

```
List Name:          LongDistance
Restriction Codes:  91___ ____ ____ ____ ____ ____ ____ ____ ____
Permission Codes:   91416 911_ ____ ____ ____ ____ ____ ____ ____
```

Permission codes are exceptions to the more general rules dictated by the restriction codes. In this example, all calls beginning with “91” are not allowed, *except* for those beginning with “91416” and “911”.

Permission codes that are shorter than a restriction code but which match a subset of such a code are not restricted. For example, if “1614” is a restricted code, the DN 161 is not restricted. In this example, calls beginning with “91” (long-distance calls) are restricted. However, calls beginning with “9” (followed by a digit other than “1”) are permitted. Therefore, local calls would be permitted in this example. (To create a restriction/permission table that restricts local calls, but not long distance calls, you would enter “9” as a restriction code and “91” as a permission code.)

When a number is dialed, the system checks the restriction and permission codes to see if the number is allowed. The following actions are performed in the order described below:

1 The DN is compared to the restriction codes. If the dialed DN is preceded by or equal to a restriction code, the DN is compared to the permission codes to see if it is an exception.

  If the DN is not restricted, or if it is an exception, the DN is called.

2 The restricted DN is compared to the permission codes. If it is preceded by or equal to a permission code, the DN is dialed. If it is not preceded by or equal to a permission code, the DN is not dialed.

When a call is not permitted, the user hears a system message indicating that the number can’t be reached from the service.

Figure 8-6xxx
Voice Security Options screen

```
                         Voice Administration

  Voice Security Options

  Maximum Invalid Logon Attempts Permitted per session:  __3

  Maximum Invalid Logon Attempts Permitted per mailbox:  __3

# Maximum Days Permitted Between Password Changes:       60

# *Password Expiry Warning (days):                        5

# *Minimum Number of Password Changes before Repeats:    5

# Minimum Password Length:                                4

# External Logon:                                         Enabled

# Call Answering/Express Messaging Thru_Dial   None On_switch [Local]
  restriction/permission codes:                 Long_Distance_1
                                                Long_Distance_2

  List Name:          On switch
  Restriction Codes:  90___ 60___ _____ _____ _____ _____ _____ _____
  Permission Codes:   90123 60245 _____ _____ _____ _____ _____

  List Name:          Local
  Restriction Codes:  91___ 90___ 60___ _____ _____ _____ _____
  Permission Codes:   _____ _____ _____ _____ _____ _____ _____

  List Name:          Long Distance 1___
  Restriction Codes:  91___ 90___ 60___ _____ _____ _____ _____ _____
  Permission Codes:   90123 60245 91416 _____ _____ _____ _____ _____

  List Name:          Long Distance 2___
  Restriction Codes:  90___ 60___ 91___ _____ _____ _____ _____ _____
  Permission Codes:   _____ _____ _____ _____ _____ _____ _____ _____

 Select a softkey >
   Save      Cancel
```

* These fields are displayed only if Maximum Days Permitted Between Password Changes is greater than 0.

\# These fields are not displayed if Call Answering (VMUIF) is installed.

The following fields are displayed:

- ***Maximum Invalid Logon Attempts Permitted per session*** - This field determines the maximum number of times that a user can make an invalid logon attempt within a single session (this limit also applies if the user tries to log on to a number of different mailboxes).When this maximum is reached within one session, the session will be terminated. You may enter a value from 1 to 99. The default is “3”.

- ***Maximum Invalid Logon Attempts Permitted per mailbox*** - This field specifies the maximum number of unsuccessful logon attempts allowed for each mailbox (this is a cumulative number). When the limit is reached, the mailbox is disabled and the user is not able to log on. To re-enable the user's mailbox, go to the Modify Local Voice User screen for that mailbox and enable the *Logon Status* field. The range is from 1 to 99. The default is "3".
- ***Maximum Days Permitted Between Password Changes*** - This field is not applicable if Call Answering (VMUIF) is installed on the system. This field determines the maximum number of days allowed between password changes. If you do not want users to have to change their passwords, set this field to "0. If this field is set to a non-zero value, users who do not change their password in the specified time, will not be able to log on to their mailbox. (The current password expires after the exact number of days specified in this field, including partial days.) To re-enable a user's mailbox, go to the Modify Local Voice User screen for that mailbox and enable the *Logon Status* field. The valid range is from 0 to 90. The default is "0".
- ***Password Expiry Warning (Days***) - This field is not applicable if Call Answering (VMUIF) is installed.. Furthermore, this field is displayed only if the *Maximum Days Permitted Between Password Changes* field is set to a value greater than "0". The value you enter determines the number of days advance notice given to a user before their password expires. The range is from 0 to 60. The default is "5".
- ***Minimum Number of Password Changes before Repeats*** - This field is not applicable if Call Answering (VMUIF) is installed. This field appears only when the field, *Maximum Days Permitted Between Password Changes*, is not "0". This number determines the number of password changes required before the same password can be re-used. The range is from 0 to 5. The default is "5".
- ***Minimum Password Length*** - This field is not applicable if Call Answering (VMUIF) is installed (see note below). This field determines the minimum number of digits required in passwords that are entered from a telephone keypad. This includes mailbox passwords, the access password used to restrict access to voice menu applications and the update password used to update voice menu applications from a DTMF phone set. It does not include the administration password that is entered when logging on at the administration terminal. If this value is set to "0", passwords are not required. The default is "4". The maximum password length is 16 digits.

Note for VMUIF subscribers: Subscribers can have passwords (from 0 digits - no password - to 16 digits in length). If a subscriber has a password (of at least one digit in length) he or she can log on to DMS Voice Mail to listen to messages from a remote phone (for example, if the subscriber is sent a remote notification). Without a password, subscribers can only log on from their “home phones”.

- ***External Logon*** - This is a read-only field. It is only applicable if Voice Messaging is installed. When it is “Enabled”, access to Voice Messaging from an external trunk is allowed. This feature can be disabled for security reasons, however, once disabled, access from external trunks is permanently revoked. The default is “Enabled”.

  Note: External logon can only be disabled by field service representatives. Once disabled, the feature can not be re-enabled.

- ***Call Answering/Express Messaging Thru-Dial Restriction/Permission codes*** - Select the restriction/permission table that will apply to call answering thru-dial and express messaging thru-dial. (This field is not applicable if Call Answering is installed.) The selection made here affects all users in the system. (Restriction/permission codes are specified in the fields below.)

  Call answering and express messaging thru dial allows callers who are connected to DMS Voice Mail during call answering or express messaging sessions to place calls by pressing “0” followed by an extension DN or an external phone number. This can become a crucial security hole in your system if restriction codes are not put in place to prevent callers from placing calls which will be charged to your organization.

  The four choices displayed in Figure 8-6 (On_switch, Local, Long distance 1, and Long distance 2) are the default names and may be different on your system.

- ***List Names and Codes*** **-** This is where you define the restriction/permission tables that are applied to various features such as custom revert and remote notification. You may create up to four sets of dialing codes (also referred to as restriction/permission tables). Each table must have a list name associated with it and can contain up to 10 restriction codes and 10 permission codes. The default names are “On Switch”, “Local”, “Long distance 1”, and “Long distance 2”. You may change these names but the name field can not be left blank. The default names suggest the types of access codes you may want to group together. The “On Switch” set may be used to restrict certain extensions on the switch. The “Local” group may be set up to restrict certain external non long distance phone numbers. “Long distance 1” and “Long distance 2” may include area codes that you wish to restrict. For example, Feature 1 (thru-dialers) might need different long distance restrictions than Feature 2 (custom revert DN).

  If you change the restriction codes so that a user’s target Remote Notification DNs are rendered invalid, Remote Notification is disabled for that user until the user’s target DN is changed.

Procedure 8-3xxx
Setting Voice Security Parameters

Starting point : Voice Administration menu, <2> entered.

1 The Voice Security Options screen appears (Figure 8-6).

2 Move the cursor to the field you wish to modify; make the required changes.

3 Choose step 3a to save the changes or 3b to cancel.

a. Use [Save].

The changes are saved and you are returned to the Voice Administration menu.

b. Use [Cancel].

Changes are discarded. The Voice Administration menu reappears.

Voice System Configuration/Voice Menu Applications Administration

If you have the Voice Menu feature package, configuration of the voice services provided to users and the administration of Voice Menu Applications are both done through one menu: the VS Config/Menu Applications Admin menu. Otherwise, only the Voice System Configuration option will be shown. See Figure 8-8 on page 8-43.

The following items are used in the configuration of voice services.

- ***Channel Allocation Table*** - in which you specify the UCD queue DNs and services of the voice channels which connect DMS VoiceMail to the switch.
- ***Voice Services-DN Table*** - in which you specify the voice services available on your system as well as the Directory Numbers (DNs) through which users gain access to these services.
- ***Voice Services Profile*** - in which you specify the broad operational parameters common to all voice services.

The following menu items are used in the administration of voice menu applications:

- ***Announcement Definitions*** **-** in which you define recorded announcements for playback within a voice menu, or as a stand-alone voice service.
- ***Thru-Dial Definitions*** **-** in which you define call handling services as a stand-alone service or to allow users to place calls to permitted numbers from a voice menu.
- ***Time-of-Day Controls Definitions*** **-** in which you define the activation of voice services according to time and date. Ways in which they are differentiated are business hours, off-hours, and holidays.
- ***Voice Menu Definitions*** **-** in which you define voice menus as sets of actions to be offered to the user. Each action corresponds to a key on the telephone keypad. Each voice menu can have a greeting that explains the purpose of the menu, and a second prompt played to users if a timeout condition is reached.

Voice services

The following are the different voice services available on your system:

Basic services

- ***Call Answering* -** provides call handling and message storage capabilities. This feature allows a user's mailbox to function like an answering machine. If a caller rings a user's phone when the user is away from his or her desk or on the phone, the caller is connected to the user's mailbox. The caller hears a greeting (recorded in the user's voice) and is prompted to leave a message after the tone.
- ***Voice Messaging*** - provides facilities that permit users to compose and send voice messages. This is the basic service offered in CPE environments and to Centrex customers in a CO environment. This service is provided with the Northern Telecom proprietary interface (MMUI).

  Note: When the MMUI interface is enabled, users have access to both call answering and voice messaging functionality. When the VMUIF interface is enabled, user's only have access to call answering functionality (known as simplified call answering).

- ***(Simplified) Call Answering* -** This service can be implemented in both a CO and a CPE environment. In a CO environment, Call Answering is generally provided for residential and small business subscribers. In a CPE environment, it is possible that not all users require full-featured voice messaging. For example, in a university setting, you may only want to provide voice messaging to staff mailboxes, not student mailboxes.

  Call Answering is available for both DTMF and dial-pulse sets. This service is essentially the same as "regular" call answering in terms of functionality. The difference is that call answering is offered along with voice messaging when the MMUI interface is enabled, whereas only simplified call answering is available when the VMUIF interface is installed.

- ***Express Messaging*** - allows users to directly place a message in another user's mailbox without first ringing the destination phone. Users first dial the Express Messaging directory number to indicate they want to use this service. They are then prompted for the mailbox. A personal verification (if recorded) is played to confirm they have reached the correct user and they are prompted to leave a message. This service requires Voice Messaging and the MMUI interface.

  Note for Meridian Networking users: Users can only use express messaging to deposit a message into another local voice user's mailbox. If networking is installed, express messaging can not be used to send a message to a user at a remote site, even if that user is defined on the system as a remote voice user.

- ***Outcalling (Remote Notification and Delivery to Non-Users) -*** Remote Notification allows users to be informed of new messages at a remote phone or pager. Delivery to non-users allows users to compose and send messages to people outside of the DMS Voice Mail system. Because Call Answering subscribers do not have voice messaging capabilities, delivery to non-users is not available if VMUIF is installed.

Networking services

- ***AMIS Networking -*** allows users to send and receive messages to or from users of other remote voice messaging systems that also use the AMIS protocol (which may include non-DMS VoiceMail systems). Users can also reply to the originator of an AMIS message. Predefined passwords or site information are not required in order to send, receive or reply to messages.
- ***Meridian Networking*** - is an optional proprietary networking service that is available on CPE but not CO systems. This feature allows subscribers to exchange messages with remote DMS VoiceMail sites. It provides enhanced capabilities above and beyond AMIS Networking.

Voice Menu Applications

Voice menu applications are custom call answering applications created by the administrator. They allow callers to listen to recorded information (announcements), leave messages for specific users, or place calls (thru-dialers). They can also route callers to particular services based on the time of day (business hours or off-hours) and can handle calls that are received during holidays by passing callers to the appropriate service (time-of-day controllers). Voice menu applications include the following services:

- ***Announcement Services*** - allow you to record messages that can be played back within a voice menu application, or as a stand-alone service that is directly dialable.
- ***Thru-Dial Services*** - access pre-defined DNs or user-prompted DNs that can be used within a voice menu service, or as a separate service with a directory number. Thru-dialers can be created to provide a variety of dialing options to users of DMS VoiceMail. Thru-dialers can be set up to allow Name Dialing, and can have restrictions barring users from dialing unauthorized numbers (such as long-distance access codes).
- ***Time-of-day Controls*** - allow you to control the activation of voice menu applications based on the date and time at which a call is received. This allows you to control the availability of voice services during off-hours and holidays.
- ***Voice Menus*** - allow you to create single-layered or multi-layered menus which present callers with a series of choices about the actions they can perform. A caller selects an action by pressing the key (on the telephone keypad) that corresponds to the action.

- ***Voice Prompt Maintenance*** - allows you or your delegates to modify the various prompts and greetings available in your voice menus and announcements using a telephone; see the chapter “Making recordings”.
- ***Remote Activation*** - allows you to enable or disable voice menu applications while you are off-site, through a standard DTMF telephone set.

For more information, see “Voice Menu Applications Administration” later in this chapter. To determine how many voice menu applications can be created, see the technical specifications in *the DMS VoiceMail Product Guide* (NTP 297-7001-010).

Note: Voice menus are not available if Call Answering (VMUIF) is installed on the system.

Voice Forms

Note: Voice Forms is an optional feature. It cannot be installed on systems that have the Call Answering feature (VMUIF).

- ***Voice Forms Administration*** - allows you to create applications that collect voice information from callers. An application consists of a series of questions, played in sequential order, to which callers give voice responses. It is as if callers are filling in a form over the phone. See “Voice Form Definitions” later in this chapter.
- ***Voice Forms Transcribers*** - allow you (or users that have been designated as transcribers) to retrieve the voice data collected by voice forms.

ACCESS

ACCESS is an optional software program. It uses a Unix interface to provide a development tool for creating specialized voice service applications such as banking-by-phone and order entry-by-phone where the system places orders for callers based on the caller input on a tone-generating telephone. ACCESS applications provide users with access to computer systems without the need for complicated terminals or a human intermediary. ACCESS applications can make use of the full range of voice and telephony functions that a digital voice processing system and a telephone switching system can offer. No special voice or telephone interface cards are needed as the switch and DMS VoiceMail together provide all of the necessary resources. ACCESS can be used to create applications for incoming or outgoing calls or for administrative purposes.

When a call is placed to the application, the system where a telephone caller places a call and the system provides information or places orders for the caller based on the caller’s input on a tone-generating telephone.

Note: ACCESS is available on CPE systems only.

For more information on ACCESS, refer to the following NTPs.

Meridian ACCESS Configuration (NTP 555-7001-315),
ACCESS Developer's Guide (NTP 555-7001-316),
ACCESS API Reference Manual (NTP 555-7001-317),
Voice Prompt Editor User's Guide (NTP 555-7001-318).

Configuring voice services on the DMS/SL-100

UCD (Uniform Call Distribution) allows a number of telephones connected to the DMS/SL-100 (known as *agent positions*) to share equally in answering incoming calls made to one or more voice service DN. Incoming calls are placed in a UCD call queue and presented to the available agent positions on a “first-in, first-out” basis. Available agent positions are placed in another queue, and the one which has been longest in the queue (and therefore has been idle the longest) is the first to be presented with a UCD call.

DMS VoiceMail uses UCD to receive calls from users who have dialed the directory number (DN) of a voice service (such as voice messaging, express messaging or a voice menu). The UCD agent positions correspond to the T1 channels through which DMS VoiceMail is connected to the DMS/SL-100. To the DMS/SL-100, T1 channels represent a set of “telephones” to which it can distribute calls.

Configuring the DMS/SL-100

Configuration of DMS VoiceMail voice services begins on the DMS/SL-100. Here you define the primary Uniform Call Distribution (UCD) queue. The primary queue contains the agents that handle the calls. It is the only UCD queue that is required and all other DMS VoiceMail voice services can share the agents in this queue.

> ***Note:*** The primary UCD queue must be set for the voice messaging feature in order for all call answering scenarios to be handled properly. In other words, do not associate the primary UCD queue with any other service (such as a voice menu or express messaging).

For each voice service that you want users to be able to dial directly, you will have to create either a line DN or a UCD queue. For services that will share the agents in the primary queue, create a line DN that forwards to the primary UCD queue. A line DN can be forwarded to a UCD queue using either Call Forward Universal (CFU) or Call Forward Fixed (CFF). This is how the voice service accesses the agents on your system. Once the call is forwarded to the queue, it will compete for agents along with the other services that share this queue.

If a particular service requires dedicated agents for some reason, create a UCD queue instead of a line DN. This UCD queue will have agents of its own. This is described in greater detail in the following sections.

Note: For services that are not directly dialable, you do not need to create a DN or a UCD queue. For example, you may have a multi-layered voice menu application in which a caller presses a key and is connected to another voice menu. Only the top-level voice menu requires a DN because this is the service that is actually called. Any other voice menus or announcements that are nested within this voice menu do not need their own DNs (unless they are going to be used as stand-alone services as well).

Configuring DMS VoiceMail

On the DMS/SL-100 you have a number of agents. Each agent corresponds to a T1 channel in DMS VoiceMail. When DMS VoiceMail is installed, you define the T1 channel that corresponds to each agent. This information is reflected in the Channel Allocation Table (CAT) in DMS VoiceMail. The CAT lists each agent that resides on the DMS/SL-100. For each agent on the DMS/SL-100, the following is specified: the corresponding T1 channel in DMS VoiceMail, the UCD queue to which the agent belongs, the agent DN and the service to which the agent/T1 channel is dedicated (this can be “ALL” for agents that are shared by all services, or a particular service can be specified). If you move agents from one queue to another in order to dedicate them to a particular service, you will have to modify the Channel Allocation Table to indicate the new queue to which the agent has been moved.

When you create a DN (or UCD queue) for a voice service on the DMS/SL-100, you will have to define a corresponding voice service DN in DMS VoiceMail. This is the number that users dial to access the service. These DNs are defined in the VSDN table. If you created a DN for the service, you will enter the DN of the line in this table. If you created a UCD queue, you will enter the UCD DN. Therefore, the line DN or UCD DN that is configured on the DMS/SL-100 becomes the *Access DN* or *Service DN* of the voice service in DMS VoiceMail.

By configuring a DN for each voice service, users will dial a different access DN to use each different service. For example, users dial one number to use voice messaging, another to use express messaging, and another one to access a voice menu. By having different DNs for each service, the services are distinguished from each other. This is important because it ensures that the proper prompts are played for the service requested. For example, when a user dials the voice messaging DN, a prompt is played asking the user to enter their mailbox number and password. However, when a user dials the express messaging DN, a prompt is played asking the user to enter the mailbox number they want to reach. When they enter the mailbox number, a personal verification is played followed by a prompt to leave a message.

Guidelines for configuring voice services

Whenever you create a new voice service you will have to decide if that service is going to share the agents in the primary voice messaging queue, or if you are going to dedicate one or more agents to that service. Most of the voice services that you create will share the agents in the primary voice messaging UCD queue.

In general, when services share the agents in the voice messaging queue, the T1 channels are used more efficiently. When T1 channels are dedicated to a particular service, the overall efficiency of the system may be reduced for the following reasons:

1. A T1 channel that is dedicated to a particular service cannot be used by Outcalling features (Remote Notification and Delivery to Non-Users) to place calls.
2. When a T1 channel is dedicated to Outcalling, this service can only use the dedicated ports to make outbound calls (i.e., it cannot use a T1 channel configured for “ALL” services.)

For example, your system has 12 T1 channels. Ten of them are shared by all services (and belong to the primary voice messaging queue). You dedicate two of your T1 channels to the Outcalling feature. When the voice messaging service or the networking services places a call, any of the 12 T1 channels can take the call. However, a remote notification call or a delivery to a non-user can only use one of the two T1 channels dedicated to Outcalling (because of the second restriction).

When a T1 channel is dedicated to a service, incoming calls from the DMS/SL-100 are still accepted on that T1 channel. This is because the DMS/SL-100, not DMS VoiceMail, is in control of incoming calls. You can, however, prohibit incoming calls on a dedicated T1 channel. For example, you can prohibit other services, such as voice messaging and voice menus, from using the T1 channels dedicated to Outcalling. This is described in the section “The Channel Allocation Table” on page 8-44.

Use the following guidelines to help you decide whether you need to dedicate T1 channels to a service:

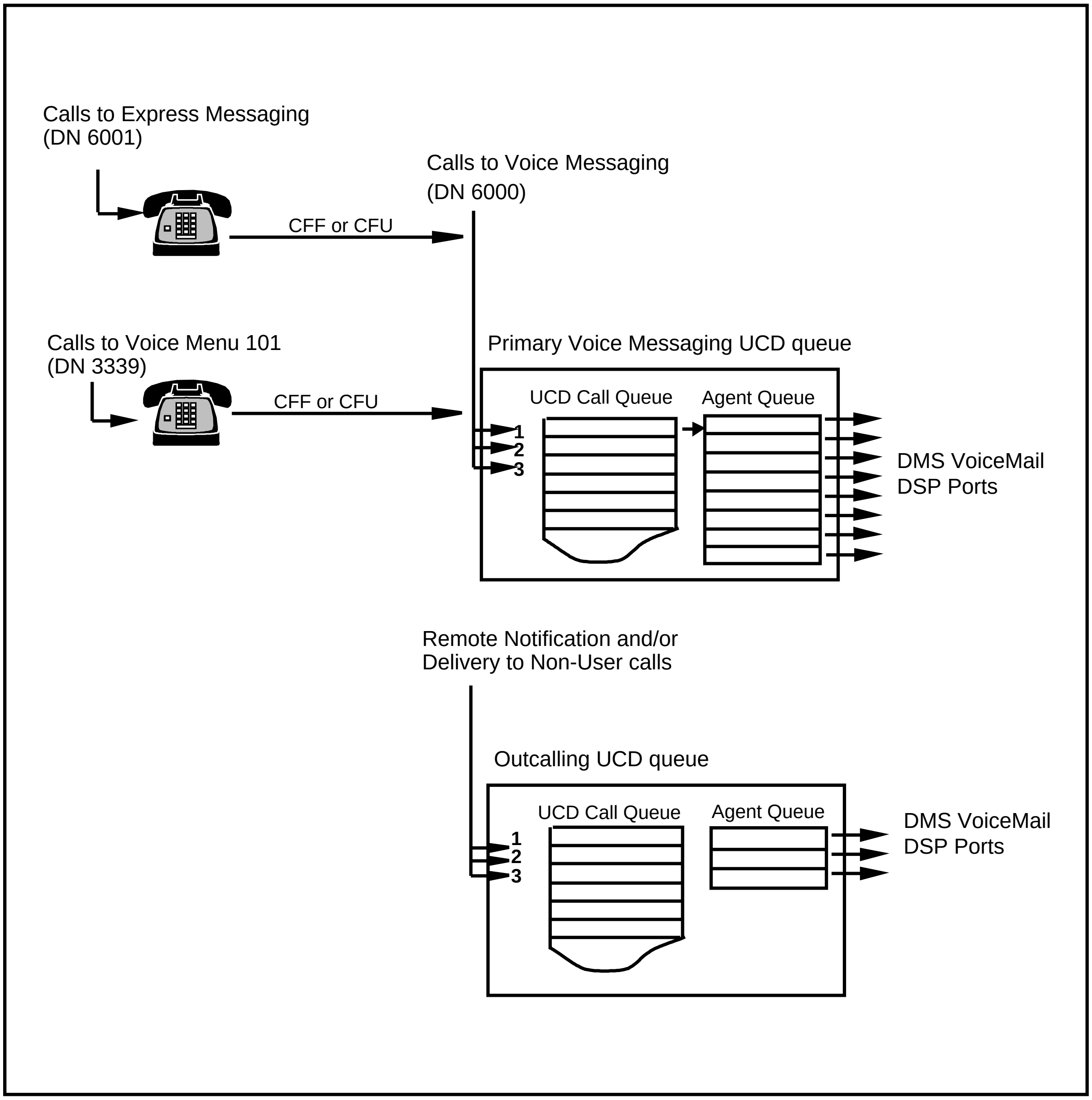
- It is crucial that the Outcalling service (which includes the remote notification and delivery to non-users features) always have access to channels. For example, in a hospital it might be very important that channels always be available for the remote notification service to guarantee that doctors will be paged if they have received urgent messages while away from their office. In this case you would not want the remote notification service to have to compete with all other services as it is urgent that doctors are notified immediately.
- Traffic studies have shown that a particular service is used a lot and that calls are being lost because the service has to compete with other services.

Other services will not be able to use these dedicated channels. Therefore, do not dedicate channels unless it is absolutely necessary. The more channels that are dedicated to certain services, the greater the possibility that there will be a noticeable degradation in the performance of other services that must share a smaller number of channels.

- Outbound ACCESS applications are crucial and it is important that an agent always be available.
- If you are only going to be using one voice service in addition to voice messaging, it is recommended that you dedicate T1 channels to it.

In Figure 8-7, line DNs have been created on the DMS/SL-100 for the express messaging service and the voice menu service. A UCD queue with dedicated agents has been configured for the Outcalling service.

Figure 8-7xxx
Voice services configuration on the DMS/SL-100



Shared configuration

For each new voice service that will share the agents in the voice messaging queue, you must configure a real or fictitious telephone set on the DMS/SL-100 and then forward it to the primary voice messaging UCD queue. To do this:

1. Create a line DN on the DMS/SL-100.
2. Call forward the DN (using CFF or CFU) to the primary Voice Messaging UCD queue.
3. The DN of the line is the DN of the new voice service. Enter this DN in the VSDN Table in DMS VoiceMail.

Step-by-step procedures are given in the section “Detailed configuration procedures”, Procedure 8-4. If you are adding new agents to the voice messaging queue, also see Procedure 8-6.

In Figure 8-7, the express messaging service and the voice menu service are configured as DNs on the DMS/SL-100 that forward to the primary voice messaging queue. These three services share the agents in the primary voice messaging queue.

Dedicated configuration

If a voice service requires dedicated T1 channels you will have to create a UCD queue. This requires that you remove some agents from the primary voice messaging queue and place them in a new UCD queue. If you have purchased additional channels (UCD agents) and added them to the system, add them to the new service queue.

If a voice service requires dedicated channels:

1. Determine the number of agents required for the given voice service.
2. Create a UCD queue on the DMS/SL-100.
3. Add agents to the new queue. These can either be moved from the voice messaging queue, or added to the new queue (if a channel expansion package has been purchased).
4. If you have moved agents from the voice messaging queue to the new service queue, go to the Channel Allocation Table in DMS VoiceMail. For each agent that now resides in the new UCD queue, enter the UCD DN of the new service queue as the Primary DN and specify the service to which the agent is dedicated.
5. The DN of UCD queue (the UCD DN) is the directory number of the new voice service. Enter this DN in the VSDN Table in DMS VoiceMail.

Step-by-step procedures are given in the section “Detailed configuration procedures”. Procedure 8-7 describes how to configure a UCD queue for the new service and move existing agents from the voice messaging queue. Procedure 8-8 describes how to add new UCD agents.

In Figure 8-7, the outcalling service is configured as a UCD queue on the DMS/SL-100.

Detailed configuration procedures

If the UCD queue for the voice messaging service has not been created on the DMS/SL-100, you will have to do this before you carry on with any of the following procedures. The UCD queue is configured using the ucdgrp table. Use the dnroute (BCS 32 or up) or wrdn table (BCS 31 and earlier) to define the DN for this UCD queue. See the *Translations Guide* (NTP 297-7001-310) for details.

Configuring a voice service that shares agents in the primary voice message queue

Follow Procedure 8-4 to configure a voice service to share the agents in the primary voice messaging queue.

If you will also be adding new agents to the voice messaging UCD queue (if you have purchased a channel expansion package), see Procedure 8-6.

Procedure 8-4xxx
Configuring a new service to share agents

DMS/SL-100 Configuration (At the MAP terminal)

1 Enter **servord** followed by <Return>. Respond to the prompts as indicated in Table 8-1.

Note: Use either the CFF (Call Forward Fixed) option or the CFU option (Call Forward Universal) to forward the DN to the voice messaging queue. CFF is recommended since it is much easier to implement.

Table 8-1xxx
Defining a DN for a voice service

Prompt	Input	Comments
SO:	NEW	
SONUMBER:	___________ $	Current date and time
DN: *	___________	Directory Number of the line
LCC:	IBN	Line class code of service
GROUP	___________	Name of the IBN customer group to which the line belongs
SUBGRP:	___________	Subgroup number
NCOS:	___________	Network class of service
SNPA:	___________	Serving NPA of the DN
LEN:	___________	Line equipment number of the line

Prompt	Input	Comments
OPTION:	CFB	Call Forward Busy
CFBCNTL:	N	(Normal assignment for CFB)
CFBDN:	xxxxxxx	The Voice Messaging UCD DN
OPTION:	CFD	Call Forward Don't Answer
CFDCNTL:	N	(Normal assignment for CFB)
CFDDN:	xxxxxxx	The Voice Messaging UCD DN
OPTION:	CFF **	Call Forward Fixed
CFFDN:	xxxxxxx	The Voice Messaging UCD DN
OPTION	CFU **	Call Forward Universal
OPTION:	$	

* *The DN of the line becomes the directory number of the new service.*

** *Choose one of CFF or CFU.*

Note: If you are using CFU, additional configuration is necessary. Go to Procedure 8-5 now. Once completed, return here and resume with the DMS VoiceMail configuration.

DMS VoiceMail Configuration

1 Log on at the DMS VoiceMail administration terminal.

2 Select Voice Administration from the Main Menu.

3 Select Voice System Configuration/Voice Menu Applications Administration.

4 Select Voice Services-DN Table.

5 Press the [Add] softkey to access the Add DN Information screen.

6 Enter the DN of the line in *Access DN* field.

7 Specify the service in the *Service* field.

8 Save the DN information.

Note: See the section "Adding DN information" on page 8-53 for more information about adding DNs.

Procedure 8-5xxx
Configuring the CFU (Call Forward Universal) option

DMS/SL-100 Configuration (At the MAP Terminal)

> ***Note:*** This procedure must be carried out for every line that forwards to the voice messaging UCD queue.

1 Use **table cfx** to define the CFU DN. This is the UCD DN of the voice messaging queue to which the voice service DN will forward. Respond to the prompts as indicated in Table 8-2.

Table 8-2xxx
Defining the CFU DN

Prompt	Input	Comments
TABLE:CFX	pos x x x x x 0 (for example: pos 4 1 9 16 0)	Where xxxx is the Line Equipment Number (LEN) of the line for the service you defined in Table 8-1 (enter a 0 at the end of the LEN)
	cha	To indicate that you want to change the DN to which CFU forwards
CFUIFDN	xxxxxxx	Enter the UCD DN of the primary voice messaging service

At a telephone set

1 Connect a phone to the line.

2 Go off hook.

3 Call forward the line to the voice messaging UCD DN.

a. Dial the call forward activation code followed by the UCD DN.

For example: *80 2326050

If you do not know what the code is, look it up in Table IBNXLA first. Check the entry for CFW. If there is no entry, configure a code. This table is described in the Installation Guide *(NTP 555-70x1-210.)*

b. Listen for the confirmation tone. This indicates that the line has been forwarded.

Important: If the DMS/SL-100 is rebooted, steps 1 to 3 will have to be repeated for each service that CFUs to the voice messaging UCD queue.

Adding new UCD agents

Use Procedure 8-6 to add new UCD agents to the primary voice messaging UCD queue. To add new UCD agents you need to purchase the channel expansion package. Each additional channel corresponds to an additional UCD agent.

Channel expansion is described in the “Hardware Modification” chapter in the *System Installation and Modification Guide* (NTP 297-7001-504).

Procedure 8-6xxx
Adding new UCD agents to the voice messaging UCD queue

DMS/SL-100 Configuration (At the MAP Terminal)

1 Check the UCDGRP table for the voice messaging queue. Specifically, check the MAXPOS (the maximum number of UCD agents that can be active in the queue). If the number of existing agents plus new agents is greater than the MAXPOS value, increase MAXPOS to support the new agents.

2 Enter **servord** followed by <Return>. For each new UCD agent, respond to the prompts as indicated in Table 8-3.

Table 8-3xxx
Adding new UCD agents

Prompt	Input	Comments
SO:	NEW	
SONUMBER:	____________ $	Current date and time
DN:	____________	Directory Number of the UCD agent
LCC:	IBN	Line class code of service
GROUP:	____________	Name of the IBN customer group to which the line belongs
SUBGRP:	____________	Subgroup number
NCOS:	____________	Network class of service
SNPA:	____________	Serving NPA of the DN
LEN:	____________	Line equipment number of the line
OPTION:	COD	Cut-off on Disconnect
OPTION:	UCD	Uniform Call Distribution
OPTION:	DGT	Digitone
OPTION:	CNF C06	6-party conferencing
OPTION:	SMDI	Simplified Message Desk Interface

Prompt	Input	Comments
LINE_NO:	___________	Line number position in the UCD SMDI group. This corresponds to the Agent ID (AI) in DMS VoiceMail. The AI and LINE_NO must match. The AI is configured in Hardware Modification at the Tools level.
UCDGRP:	___________	The name of voice messaging UCD queue (UCDNAME from the UCDGRP table)
AUTOLOG:	Y	Autologon capability required
OPTION:	$	

DMS VoiceMail Configuration

1 Program DMS VoiceMail to recognize the new channels (agents). See the “Hardware Modification” chapter in the *System Installation and Modification Guide* (NTP 297-7001-504) for details. During this procedure you will specify which T1 channel corresponds to which agent, and whether the channel is shared or dedicated to a particular service. The Channel Allocation Table will be updated automatically to reflect these changes.

Dedicating agents to a voice service

Use the following procedure to dedicate one or more agents to a new voice service. If you are also adding new agents, see Procedure 8-8.

Procedure 8-7xxx
Dedicating agents to a voice service

DMS-100/SL-100 Configuration

1 At the MAP terminal, enter **table ucdgrp** followed by <Return> to configure the UCD queue. Respond to the prompts as indicated in the *Translations Guide* (NTP 297-7001-310). For the MAXPOS prompt, be sure to enter a value equal to or greater than the number of agents that you will be transferring or adding to this queue.

2 Use **table dnroute** (BCS 32 and up) or **wrdn** (BCS 31 and earlier) to define the directory number (DN) of the new UCD queue. Respond to the prompts as indicated in the *Translations Guide* (NTP 297-7001-310).

3 Enter **servord** followed by <Return> to move UCD agents from the voice messaging UCD queue to the new UCD queue. Respond to the prompts as indicated in Table 8-4.

Table 8-4xxx
Moving a UCD agent

Prompt	Input	Comments
SO:	CHF	
SONUMBER:	__________ $	Current date and time
DN_OR_LEN:	__________	DN or Line equipment number of the UCD agent
OPTION:	SMDI	Simplified Message Desk Interface
LINE_NO:	__________	Line number position in the UCD SMDI group. This corresponds to the Agent ID (AI) in DMS VoiceMail. The AI and LINE_NO must match. The AI is configured in Hardware Modification at the Tools level.
UCDGRP:	__________	Name of the new service UCD queue to which the agent belongs (UCDNAME from table UCDGRP)
AUTOLOG:	Y	Autologon capability required
OPTION:	$	

DMS VoiceMail Configuration

1 Log on at the DMS VoiceMail administration terminal.

2 Select Voice Administration from the Main Menu.

3 Select Voice System Configuration/Voice Menu Applications Administration.

4 Select Voice Services-DN Table.

5 Press the [Add] softkey to access the Add DN Information screen.

6 Enter the UCD DN (that was configured in the wrdn table) in the *Access DN* field.

7 Specify the service in the *Service* field.

8 Save the DN information.

Note: See the section “Adding DN information” on page 8-53 for more information about adding DNs.

9 Return to the Main Menu.

10 Select System Status and Maintenance from the Main Menu.

11 Select T1 Channel Status.

12 Disable the T1 channels that you have just added (see page 10-13 for more information).

13 Return to the Main Menu.

14 Select Voice Administration.

15 Select Voice System Configuration/Voice Menu Application Administration.

16 Select Channel Allocation Table. For each agent that has been moved:

a. Modify the *Primary DN* field. Enter the UCD DN of the new UCD queue that you configured in the dnroute or wrdn table in step 2.

b. In the *Channel DN* field, enter the 7-digit directory number of the corresponding UCD agent (see Table 8-3).

c. In the *Service* field, specify the service to which the agent is dedicated.

See the description of the Channel Allocation Table on page 8-44 for details.

17 Reboot the DMS VoiceMail system for the changes made to the CAT to take effect.

Adding new UCD agents

If you have purchased additional channels (UCD agents), follow Procedure 8-8 to add them to the DMS/SL-100. You do not have to modify the Channel Allocation Table after adding new agents since the Hardware Modification procedure does this automatically.

Procedure 8-8xxx
Adding new UCD agents

DMS/SL-100 Configuration (At the MAP Terminal)

1 Check the UCDGRP table for the queue(s) to which you will be adding new agents. Specifically, check the MAXPOS (the maximum number of UCD agents that can be active). If when you add the new agents to the existing agents, the number of agents exceeds the MAXPOS value, you will have to increase it to support the new agents.

2 Enter **servord** followed by <Return>. For each new UCD agent, respond to the prompts as indicated in Table 8-5. If you don’t want to add all of the new agents to the new service queue, add the remainder to the voice messaging queue.

Table 8-5xxx
Adding new UCD agents

Prompt	Input	Comments
SO:	NEW	
SONUMBER:	__________$	Current date and time
DN:	__________	Directory Number of the UCD agent
LCC:	IBN	Line class code of service
GROUP:	__________	Name of the IBN customer group to which the line belongs
SUBGRP:	__________	Subgroup number
NCOS:	__________	Network class of service
SNPA:	__________	Serving NPA of the DN
LEN:	__________	Line equipment number of the line
OPTION:	COD	Cut-off on Disconnect
OPTION:	UCD	Uniform Call Distribution

Prompt	Input	Comments
OPTION:	DGT	Digitone
OPTION:	CNF C06	6-party conferencing
OPTION:	SMDI	Simplified Message Desk Interface
LINE_NO:	__________	Line number position in the UCD SMDI group. This corresponds to the Agent ID (AI) in DMS VoiceMail. The AI and LINE_NO must match. The AI is configured in Hardware Modification at the Tools level.
UCDGRP:	__________	Name of UCD queue to which the agent belongs (UCDNAME from table UCDGRP)
AUTOLOG:	Y	Autologon capability required
OPTION:	$	

DMS VoiceMail Configuration

1 Program DMS VoiceMail to recognize the new channels (agents). See the “Hardware Modification” chapter in the *System Installation and Modification Guide* (NTP 297-7001-504) for details. During this procedure you will specify which T1 channel corresponds to which agent, and whether the channel is shared or dedicated to a particular service. The Channel Allocation Table will be updated automatically to reflect these changes.

The VS Config/Menu Applications Admin menu

The Voice System Configuration/Voice Menu Applications Administration menu is displayed when you select item <3> from the Voice Administration Menu (see Figure 8-8). Performing voice system configuration involves using the Channel Allocation Table, the Voice Services Profile, and the Voice Services-DN Table. Voice menu applications administration involves creating service definitions for announcements, thru-dialers, time-of-day controllers and voice menus. If the Voice menu applications feature is not installed, access to voice menu applications administration will not appear and the selection will simply read “Voice System Configuration”.

> ***Note:*** If you have not reviewed the information on Configuring Voice Services, you should do so prior to using the Voice System Configuration options in this menu.

Figure 8-8xxx
The VS Config/Menu Applications Admin menu

```
                  VS Config/Menu Applications Admin

             1     Channel Allocation Table
             2     Voice Service-DN Table
             3     Voice Services Profile
           * 4     Announcement Definitions
           * 5     Thru-Dial Definitions
           * 6     Time-of-Day Control Definitions
           * 7     Voice Menu Definitions

 Select an item >
    Exit
```

* These options only appear if the Voice Menu feature is enabled.

Each item in this menu is described in detail in the following sections.

Voice System Configuration

Voice system configuration includes associating T1 channels in the DMS VoiceMail system with agents in the switch, assigning DNs to voice services, and configuring general parameters that affect all voice services. The first two items in the VS Config/Menu Applications Administration menu (Figure 8-8) are used in voice system configuration.

The Channel Allocation Table

The Channel Allocation Table (CAT) (Figure 8-11) should only be configured by those who are knowledgeable about programming the switch. Normally, you will not have to configure this table. The technician that installed the DMS VoiceMail system should have done so in order to ensure consistency between the switch and DMS VoiceMail. After installation, it may be necessary to alter this table at a later date if, for example, it is decided that certain services require dedicated channels for effective performance or if channels are added to the system.

The Channel Allocation Table (CAT) (Figure 8-11) associates each agent on the switch with a voice channel (T1 channel) on a DMS VoiceMail T1 card. Agents are identified by a Terminal Number (Routing Address) and Directory Number. Each T1 channel must be associated with an existing UCD agent in the Central Office switch (e.g., DMS-100) or SL-100 data base to handle the queuing of calls coming in to DMS VoiceMail and to handle dial-out features such as remote notification and delivery to non-users.

A channel may be shared by all services or dedicated to a specific service. Dedicated channels may reduce the overall efficiency of the system since dedicated channels currently not in use can’t be used by the Outcalling service (namely, Remote Notification and Delivery to Non-Users). Also, when a channel is dedicated to the Outcalling service, Outcalling features are restricted to those channels (i.e., they can not use a channel configured for “ALL” services). Therefore, most of your channels should be shared by all services. However, certain features may require dedicated channels. See page 8-31 for guidelines.

When a channel is dedicated to a service in the CAT, incoming calls from the switch are still accepted on that channel. This is because the switch, not DMS VoiceMail, is in control of incoming calls. For each dedicated channel for which you want to prohibit incoming calls, make sure that there is one agent that has not been assigned to any UCD queue.

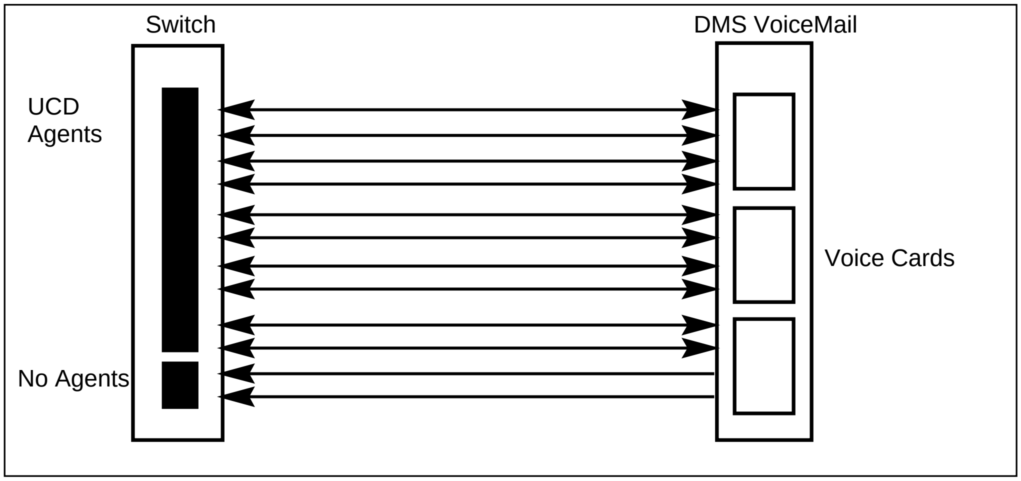
For example, your system has a total of 12 agents. To dedicate two channels to outcalling so that those channels do not accept incoming calls:

1 In the switch, only assign a maximum of 10 agents to your UCD queues.

2 In the Channel Allocation Table, make sure that the two outcalling channels are associated with the two agents that were not assigned to a queue. (You can enter the DN of the primary voice messaging UCD queue as normal. It is ignored when the TN or routing address corresponds to an agent that is not assigned to a queue.)

In Figure 8-9 the top ten channels are set to “ALL” in the CAT and the bottom two channels are reserved as outgoing channels for the Outcalling service.

Figure 8-9xxx
Prohibiting incoming calls on dedicated channels



To get to the Channel Allocation Table you will need to specify the T1 link number for the Channels you wish to view/modify.

When you select the Channel Allocation Table item from the VS Config/Menu Applications Admin menu, you are first presented with a screen from which you must specify the T1 link for which you want to view the CAT (see Figure 8-10).

Figure 8-10xxx
The Selectable Channel Allocation Table data menu

```
                    VS Config/Menu Applications Admin

Channel Allocation Table

                         Primary Link             Secondary Link
T1 Link Number           (Node-Card-Span)         Node-Card-Span)
       1                 13-1-1                   14-3-1
       2                 13-1-2                   14-3-2

Move the cursor to the item and press the spacebar to select >
   Exit        View/Modify
```

Figure 8-11xxx
The Channel Allocation Table

```
                    VS Config/Menu Applications Admin


Channel Allocation Table      Primary Link: 13-1-1     Secondary Link: 14-3-1

Choice of Services:
ALL  All Services          ACC  Meridian ACCESS*    AN   AMIS Networking
AS   Announcement Service  EM   Express Messaging   GS   Greetings Service
NW   Meridian Networking*  OC   Outcalling Agent    PM   Prompt Maintenance
RA   Remote Activation     TS   Thru-Dial Service   TR   Transcription Service
VA   Voice Administration  VF   Voice Forms Service MS   Voice Menu Service
VM   Voice Messaging


Channel           Routing Address  Primary DN     Channel DN     Service

   1              63-1              3650           2800           ACC  Class: ___
   2              63-2              3650           2801           ALL
   3              63-3              3650           2802           ALL
   4              63-4              3650           2803           ALL
  ...             ...               ...            ...            ...
  24              63-1              3650           2804           ALL




Select a softkey >

     Save              Cancel
```

*   These features are only available on CPE systems.

Note: To change the Primary DN and/or Channel DN you must first disable the T1 channel. Channels are disabled from the T1 Channel Status screen. This screen is accessed from the System Status and Maintenance menu. Rows that are displayed in bold type on the CAT represent disabled channels. Modifiable fields are underlined. The Routing Address can only be modified from the Tools menu (Modify Hardware) described in Appendix A, “System Administration Tools”.

The following fields are displayed in the Channel Allocation Table:

- ***Primary Link*** - This is the location of the primary T1 link in the system. This represents the node-card-span location.
- ***Secondary Link*** - This is the location of the secondary or backup T1 link in the system. This represents the node-card-span location.

- ***Choice of Services*** - This is a list of voice services and their associated acronyms. See "Configuring voice services on the DMS/SL-100" earlier in this chapter for information on shared and dedicated channel configuration. Typically, most of your channels will be shared (i.e., ALL is specified).

  The following services are feature-dependent: Greetings Service (Call Answering), Express Messaging (Voice Messaging), Meridian Networking, Meridian ACCESS, AMIS Networking, Outcalling Agent, Transcription Service (Voice Forms) and Voice Forms Service. Announcement Service, Thru-Dial Service, Voice Menu Service, Remote Activation, and Prompt Maintenance are available only if voice menus are installed. Any of the above features that are not installed on your system will not be displayed in the CAT.

- ***Channel*** - The channel number. This is a read-only field.
- ***Routing Address*** - This is a read-only field specifying the location of the corresponding agent in the switch. This is the Message Desk Number. The elements in the address represent the message desk number and the terminal number and is expressed in the format xx-yyyy.
- ***Primary DN*** - This is the directory number assigned to the agent queue that contains this agent (channel). On CO systems this is typically the 7-digit directory number. On CPE systems, this is usually a 4- or 5-digit number.

  Note: On CO systems for which the SMDI link is set to 10-digit messaging, enter the full 10-digit DN (including the area code).

  For channels that are shared by all services, this is the DN of the primary voice messaging queue. However, if the channel is dedicated to a particular service, enter the DN of the corresponding service UCD queue (as configured in the dnroute or wrdn table).

  Note: The channel must be disabled before changing the Primary DN. If the T1 channel is not disabled, this is a read-only field.

- ***Channel DN*** - This is the DN of the UCD agent that corresponds to this channel. This field cannot be modified unless the T1 channel is disabled.
- ***Service*** - The DMS V oiceMail service to which the channel and agent are dedicated. The default is "ALL", indicating a shared channel.

  Meridian ACCESS causes a second field, *Class*, to be displayed. Enter the required number for the specified Access class.

Procedure 8-9xxx
Modifying the Channel Allocation Table

Starting point : The Main Menu.

> ***Note:*** *Update the Channel Allocation Table only when the system is idle. You must first disable the channels that will be updated.*

1 Select System Status and Maintenance from the Main Menu.

2 Select T1 Channel Status.

3 Select the [Disable Channel] softkey.

You are prompted for the in-service channel number.

4 Enter the number(s) of the channel(s) you want to disable and press <Return>.

5 Select [Exit] until you have reached the Main Menu.

6 Select Voice Administration.

7 Select Voice System Configuration.

8 Select Channel Allocation Table.

The Selectable Channel Allocation Table data menu appears.

See Figure 8-10.

9 Select the Channel you wish to modify.

See Figure 8-11.

10 For each disabled channel, you may modify the Primary DN, Channel DN, and the Service if the channel is being re-allocated.

11 Choose step 11a to save the changes or 11b to cancel.

a. Use [Save].

The changes are saved and you are returned to the selectable Channel Allocation Table data menu.

b. Use [Cancel].

You are returned to the selectable Channel Allocation Table data menu.

The Voice Services-DN Table

The Voice Services-DN (VSDN) Table (Figure 8-12) lists the Directory Numbers (DNs) associated with specific voice services. A DN is required for each voice service that you want users to be able to access directly by dialing a unique DN. The VSDN Table maps voice services onto DNs so that when DMS Voice Mail receives an incoming call, it looks up the DN in the table to determine which service is being requested and which prompts to play.

For every service you plan to add to the VSDN table, an existing line DN (or UCD DN) must already be configured on the switch. See “Configuring voice services on the DMS/SL-100” earlier in this chapter.

If a voice service is going to share the agents in the voice messaging queue, you must first ensure that there is an available DN on the DMS/SL-100, or configure one if there is not. If you are going to dedicate agents to the service, you must create a UCD queue on the DMS/SL-100 for the service (if there are none available). See “Configuring voice services on the DMS/SL-100” earlier in this chapter.

At the very least you must define a DN for Voice Messaging. This is the DN that users dial to log on to DMS Voice Mail and access their mailboxes. The other DNs are essentially optional. However, the following DNs are commonly configured: at least one express messaging DN (if MMUI is enabled); if voice menus are installed, a DN for both remote activation and voice prompt maintenance as well as DNs for any directly dialed voice menu applications such as announcements, thru-dialers, time-of-day controllers, voice menus and voice forms (if installed).

Figure 8-12xxx
The Voice Services-DN Table

```
                    VS Config/Menu Applications Admin
Voice Services-DN Table


  DN *                      Service      Comment

  2663602                   NW           Networking
  2663650                   EM           Express Messaging
  2663651                   PM           Prompt Maintenance
  2663654                   VM           Voice Messaging
  2663661                   RA           Remote Activation
  2663662                   TS   2000    Thru-Dial
  2663663                   EM           Express Messaging
  2663665                   AS   2001    Voice Messaging

Move the cursor to the item and press the space bar to select >

    Exit          Add          View/         Delete
                               Modify
```

*   The entries in this table are sorted by DN.

Note: The entries in the VSDN Table are sorted by DN.

The Voice Services-DN Table includes the following read-only fields:

- ***DN*** - (Directory Number) The DN for the voice service. On CO systems, this is the 7-digit line DN or UCD DN. On CPE systems, this is usually a 4- or 5-digit number.

  Note: On CO systems for which the SMDI link is set to 10-digit messaging, enter the full 10-digit DN (including the area code).

- ***Service*** - The service that is reached when the corresponding DN is dialed.

  Voice menu applications display a corresponding ID number.

- ***Comment*** - A description of the voice service.

Procedure 8-10xxx
Adding, Modifying And Deleting Voice Service DNs

Starting point : The Voice System Configuration Menu, <2> entered.

1 The Voice Services-DN Table appears (Figure 8-12).

2 Choose step 2a to add a service DN; 2b to modify an existing service DN, 2c to delete an existing service DN, or 2d to leave the VSDN Table.

a. Use [Add].

The Add DN Information screen appears; proceed to the next section, “Adding DN information”, for details.

b. Use the cursor keys to move the cursor to the required voice service and press <Space Bar> to select it; use [View/Modify].

The View/Modify DN Information screen appears; proceed to “Modifying DN information” later in this chapter for details.

c. Use the cursor keys to move the cursor to the required voice service and press <Space Bar>; use [Delete].

The Delete DN Information screen appears; proceed to “Deleting DN information” later in this chapter for details.

d. Use [Exit].

The Voice System Configuration menu is redisplayed.

Adding DN information

The Add DN Information screen (Figure 8-13) is accessed from the VSDN Table and is used to assign available DNs to voice services.

Figure 8-13xxx
The Add DN Information screen

```
                    VS Config/Menu Applications Admin

Add DN Information

Choice of Services:
AN   AMIS Networking       AS   Announcement Service     EM  Express Messaging
GS   Greetings Service     ACC  Meridian ACCESS *        NW   Meridian Networking *
PM   Prompt Maintenance    RA   Remote Activation        TS   Thru-Dial Service
TD   Time-of-Day Controls  TR   Transcription Service    VF   Voice Forms Service
MS   Voice Menu Service    VM   Voice Messaging

Access DN:  ____________________________

Service:    _______________

Comment:    ____________________________

Select a softkey >

   Save        Cancel
```

*   These are optional features that are available on CPE systems only.

CAUTION
Entering Access DNs and Service IDs
Each Access DN, Service ID and Mailbox ID must be unique. When entering these numbers, ensure that you are not duplicating existing DNs/IDs.

The following fields are displayed:

- ***Choice of Services*** - This field lists the available voice services. The list is sorted horizontally according to the feature description, not the acronym.

- ***Access DN*** **-** This is the DN that callers dial when accessing the voice service. This is either the line DN or UCD DN as configured on the DMS/SL-100. If there are no available DNs they will have to be programmed into the switch by a technician. You must define a DN for Voice Messaging. This is the DMS VoiceMail Access Number, required by users to log on to DMS VoiceMail and access their mailboxes. The other DNs are optional.
- ***Service*** **-** This field defines which service is to be called up when the Access DN is dialed. Depending on the service selected, an extra field may be displayed. These are explained in the following descriptions.

  Note: To configure a DN for the Call Answering service, choose VM (voice messaging) as the service.

  AN This selection is possible only if AMIS networking is installed. No other fields are displayed when this service is selected.

  AS This selection is possible only if voice menus are installed. You are prompted to enter an Announcement ID. This ID is defined when you add an announcement definition. It distinguishes the announcement from all other voice services. When the access DN is dialed, the announcement associated with the ID entered in this field is played. (You do not have to define the announcement before making an entry in the VSDN table. However, if you enter an ID in this field, be sure to write it down and use it when defining the announcement.)

  EM When you specify Express Messaging, another field, *Mailbox ID*, is displayed. This is an optional field.

  It is possible to have several Express Messaging services. Express Messaging is typically used to provide users with a service whereby they can leave messages in mailboxes without actually ringing the destination phone. Do not enter a Mailbox ID for this type of service.

  You can also create Express Messaging services that connect callers to a specific mailbox. In this case you will need to enter a mailbox number in the *Mailbox ID* field. This is useful if, for example, you want to create a ‘suggestion box’. You can ask users to dial the Express Messaging DN and leave their suggestions in the mailbox. You can then play the messages back. If the mailbox number you specify has not been added to the system (through User Administration), do so after adding the Express Messaging DN.

  Each Express Messaging service you create will have a unique Access DN (make sure there are enough line DNs in the switch to accommodate a number of Express Messaging services).

Up to 18 digits can be entered in the Mailbox ID field to accommodate site and location codes for Networking, if installed.

GS This selection is possible only if Call Answering (VMUIF) is installed. This service allows subscribers to update their greetings in a manner that requires no keypad input. A DN should be created for this service to allow subscribers without digitone phones (i.e., those with rotary phones) to directly connect to the Greetings Service by dialing the specified Access DN. Once connected, the service prompts the subscriber to speak at certain times and requires no keypad input. This can also be provided to subscribers with digitone phones if they desire a simplified interface for changing greetings.

The greetings service can also be included within a voice menu. However, keep in mind that rotary phone users will not be able to access voice menus, and therefore, cannot access this service through a voice menu. To service your rotary phone subscribers you need to define a DN in the VSDN table.

ACC This selection is only possible if ACCESS is installed. ACCESS is only available on CPE systems. You are prompted to enter the Class of the ACCESS application and the revert DN (the DN to which calls are passed if the ACCESS application is unavailable).

NW This selection is possible only if Meridian networking is installed. No other fields are displayed when this service is selected.

PM Prompt Maintenance. This selection is only possible if voice menus are installed. If selected, no other fields are displayed.

RA You are prompted for a password. (This is the password required by anyone dialing the remote activation DN in order to use the service to modify voice menu applications.)

Note: If the password field is left blank, remote activation is disabled.

TS You are prompted for the ID of the thru-dialer that is to be retrieved when the access DN is dialed. This ID is configured in the Add a Thru-Dial Definition screen.

TD You are prompted for the ID of the time-of-day controller that is to be retrieved when the access DN is dialed. This is the ID that is entered in the Add Time-of-Day Control Definition screen.

TR This selection is possible only if voice forms are installed. You are prompted for the Voice Form ID of the voice form to be accessed by the DN. This is an optional field. If you enter an ID, that voice form will be automatically retrieved for the transcriber. If you do not enter an ID, the transcriber will have to enter the ID of the form he or she wants to transcribe. If you want to provide transcribers with automatic logon to particular voice forms, you will need several DNs for TR. You should also create a DN that does not reference a particular form, so that it can be used as a general access to the transcription service.

VF This selection is possible only if voice forms are installed. You are prompted for the ID of the voice form to be retrieved when the access DN is dialed.

MS You are prompted for the ID of the voice menu to be retrieved when the access DN is dialed.

VM No other fields are displayed.

- ***Comments -*** This field is optional and can be used for descriptive purposes.

Procedure 8-11xxx
Adding DN Information

Starting point ; The Voice Services-DN Table, [Add] entered.

1 The Add DN Information screen appears (Figure 8-13).

2 Move the cursor to each field and enter the required information.

3 Choose step 3a to save the changes or 3b to cancel.

a. Use [Save].

The addition is made and you are returned to the Voice Services-DN Table.

b. Use [Cancel].

The addition is not made and you are returned to the Voice Services-DN Table.

Viewing/Modifying DN Information

Once added to the system, voice service directory numbers can be modified by accessing the View/Modify DN Information screen (Figure 8-14). See the preceding section, “Adding DN information” for field descriptions.

Figure 8-14xxx
The View/Modify DN Information screen

```
                     VS Config/Menu Applications Admin

View/Modify DN Information

Choice of Services:
AN   AMIS Networking       AS   Announcement Service     EM  Express Messaging
GS   Greetings Service     ACC  Meridian ACCESS *        NW   Meridian Networking *
PM   Prompt Maintenance    RA   Remote Activation        TS   Thru-Dial Service
TD   Time-of-Day Controls  TR   Transcription Service    VF   Voice Forms Service
MS   Voice Menu Service    VM   Voice Messaging


Access DN:  _3651_______________________

Service:    EM__          Mailbox ID:      6054_______________

Comment:    Personnel Division____________
______________________________________________________________________________

    Save           Cancel
```

*   These are optional features that are available on CPE systems only.

Procedure 8-12xxx
Modifying DN Information

Starting point : The Voice Services-DN Table.

1 Move the cursor to the voice service you want to view/modify and press the <Space Bar> to select it.

2 Use the[View/Modify] softkey.

The Modify DN Information screen appears (Figure 8-14).

3 Change the information as required.

If you are changing the Access DN or a Service/Mailbox ID, make sure the new DN/ID is unique and does not duplicate an existing DN/ID.

4 Choose step 4a to save the changes or 4b to cancel.

a. Use [Save].

The changes are saved and you are returned to the VSDN Table.

b. Use [Cancel].

The changes are not saved and you are returned to the VSDN Table.

Deleting DN information

Use the Delete DN Information screen (Figure 8-15) to delete Directory Numbers from the Voice Services-DN Table. The fields on this screen are read-only.

Figure 8-15xxx
The Delete DN Information screen

```
                    VS Config/Menu Applications Admin

Delete DN Information

Choice of Services:

AS   Announcement Service    EM  Express Messaging       GS  Greetings Service
ACC  Meridian ACCESS *        NW   Meridian Networking * PM   Prompt Maintenance
RA   Remote Activation        TS   Thru-Dial Service      TD   Time-of-Day Controls
TR   Transcription Service    VF   Voice Forms Service    MS   Voice Menu Service
VM   Voice Messaging


Access DN:  56672

Service:     EM             Mailbox ID:       6054

Comment:     Personnel Division

________________________________________________________________________

OK to          Cancel
delete
```

*   These are optional features that are only available on CPE systems.

Procedure 8-13xxx
Deleting DN Information

Starting point : The Voice Services-DN Table.

1 Move the cursor to the voice service you want to delete and press the <Space Bar> to select it.

2 Use the [Delete] softkey.

The Delete DN Information screen appears (Figure 8-15).

3 Choose step 3a to delete the service or 3b to cancel.

a. Use [OK to Delete].

The entry is deleted and you are returned to the Voice Services-DN Table.

b. Use [Cancel].

You are returned to the Voice Services-DN Table without the entry being deleted.

The Voice Services Profile

The Voice Services Profile screen (Figure 8-16) allows you to set parameters that apply to all voice services (other than voice messaging services such as express messaging and call answering).

Figure 8-16xxx
Voice Services Profile screen

```
                  VS Config/Menu Applications Admin
Voice Services Profile

Voice Services Volume: 1

Timeouts
  Command Entry:            3.5_  seconds   Short Disconnect:10.0__ seconds
  Record (mm:ss):           02:00___

Maximum Prompt Size(mm:ss):02:00___ Business Hours Default:  08:30_to 17:00

Act on AMIS Initiation Tone: [No] Yes

Holidays:
     Start Date     End Date    Start Time     Comments
     (mm/dd/yy)     (mm/dd/yy)  (hh:mm)
  1  12/20/92       01/04/92    08:30          Christmas________
  2  01/18/92       ________    08:30          M.L. King________
  3  02/15/92       ________    08:30          Washington_______
  4  05/30/92       ________    08:30          Memorial Day_____
  5  07/01/92       ________    08:30          Independence_____
  6  09/05/92       ________    08:30          Labor Day________
  7  10/01/92       ________    08:30          Columbus Day_____

                                                            MORE BELOW
Select a Softkey >
    Save          Cancel
```

The Voice Services Profile includes the following fields:

- ***Voice Services Volume*** - This is a read-only field which indicates which volume contains voice menus and voice forms (if any).

  When performing a partial backup after changes have been made to any voice menu application or voice form, be sure to back up the volume indicated here in order to preserve your changes.

- ***Timeouts* -** The values you enter in the following fields determine how long the system will wait under certain conditions before the system takes action (such as disconnecting the caller from the service or playing a delayed prompt).

- ***Command Entry*** - The value entered here represents the amount of time (in seconds) that callers have to enter or complete a command before the appropriate action is taken by the service currently in operation. This field applies to voice menus, announcements and thru-dialers. The default is 3.5 seconds. The range is from 1.0 to 5.0 seconds.
- ***Short Disconnect*** - This value represents the maximum amount of time (in seconds) that the system will wait for a response before disconnecting or connecting a caller to an attendant. This value also affects callers who do not have a touch tone phone and therefore cannot enter commands to respond to prompts. The value entered here determines how long the system will wait before connecting the caller to an attendant. This field applies to voice menus, announcements and thru-dialers. The default is 10.0 seconds. You may enter a value from 1.0 to 30.0.
- ***Record*** - The length of time (in minutes and seconds) a voice service remains active while recording silence. If silence continues to this specified length of time, the voice service disconnects. The default is two minutes. You may enter a value from 00:06 to 05:00. This affects all voice services other than voice messaging and its associated features (login, call answering, express messaging).

• ***Maximum Prompt Size*** - The value entered here determines how long a voice prompt (e.g., an announcement, a menu choice) is allowed to be before the appropriate action is taken by the service currently in operation. For example, recording may stop if this maximum is reached. The default is three minutes. You may enter a value from 00:30 to 10:00.

• ***Business Hours Defaults*** - The default business hours for the system, used by time-of-day controllers; see “Time-of-Day Controls” later in this chapter for details. The defaults are “08:30 to 17:00”.

• ***Act on AMIS Initiation Tone*** - If an AMIS call comes in through a voice service DN, the voice service (such as a voice menu or announcement) will either ignore (“No”) or react to the AMIS tone and transfer the call to the appropriate AMIS agent (“Yes”). If this field is set to “No”, you will have to configure a DN specifically for the AMIS service in the VSDN table. If you plan on using a voice menu or thru-dialer to accept AMIS calls, then this field must be set to “Yes”.

  Note: If you set this field to “Yes”, make sure that the *Short Disconnect* field is set to at least 10 seconds (10 seconds is the default). This value determines how long the system will wait for a response (telephone keypad entry) before disconnecting a call. Otherwise, an AMIS call that connects to a voice menu or thru-dialer may be prematurely disconnected.

- ***Holidays*** - Up to 20 holidays can be defined. The holidays you specify here are used when defining time-of-day controls; see “Time-of-Day Controls” later in this chapter for details.
- ***Start Date*** - This field is mandatory. Specify the date on which the holiday begins. The date format follows that defined in the General Options screen, selectable from the General Administration menu.
- ***End Date*** - Specify the date on which the holiday ends (this is optional). If you enter a date, it must be later than or the same as the start date. If no end date is specified, the holiday will end on the start date. If the holiday ends on a regular day, the holiday will end at the end of the business day (five o’clock, for example). However, if it ends on a non-business day, the holiday will end at the end of the day (midnight).
- ***Start Time*** - The time at which the holiday starts on the start date. This will usually be the normal start of a business day (specified using the 24-hour clock).
- ***Comments*** - This field is optional. You may enter up to 11 characters to describe the holiday you are defining.

Procedure 8-14xxx
Setting Voice Service Parameters

Starting point : The Voice System Configuration Menu, <3> entered.

1 The Voice Services Profile screen appears (Figure 8-16).

2 Modify the existing information as needed.

3 A new holiday entry can be entered on the first available blank line; the screen can be scrolled to view additional lines.

4 Choose step 4a to save the changes or 4b to cancel.

a. Use [Save].

The changes are saved and you are returned to the Voice System Configuration menu.

b. Use [Cancel].

You are returned to the Voice System Configuration menu.

Voice Menu Applications Administration

Voice menu applications administration involves the creation and modification of menu-driven voice processing applications known collectively as *Voice Menu Applications*. Voice menu applications can be as simple as a single automated attendant, or as complicated as a multi-level series of menus. Your applications can prompt the subscriber to perform certain actions by pressing keys on the telephone keypad. These actions can, in turn, activate other voice menus, play recorded announcements, or place calls. For examples of voice menu applications, see Appendix B.

The following items in the VS Config/Menu Applications Admin menu (Figure 8-8) are relevant to voice menu applications administration:

- ***Announcement Definitions*** allow you to define recorded announcements for playback within a voice menu, or as a stand-alone voice service.
- ***Thru-Dial Definitions*** allow you to define call handling services that allow subscribers to place calls to permitted numbers from a voice menu, or as a stand-alone service.
- ***Time-of-Day Control Definitions*** allow you to determine the activation of voice services according to time and date.
- ***Voice Menu Definitions*** allow you to define sets of actions (stored in a voice menu) to be offered to the subscriber. When a caller reaches a voice menu a prompt is played which informs the caller to press particular keys on the telephone keypad in order to perform certain actions. Each voice menu can also have a greeting that explains the purpose of the menu, as well as customized prompts that are played if a timeout condition is reached or if help is explicitly requested.

Voice menu applications and associated voice recordings are administered from the administration terminal and its associated telephone. After you initially set up your voice menu applications, you can update or change voice prompts and announcements from a remote site using a touch-tone phone. This feature is referred to as Voice Prompt Maintenance. See the chapter “Making recordings” for details. This permits a greater flexibility in administration activities, since it is likely that many of the voice recordings made for a given application may require frequent updating.

> ***Note:*** If you try to gain access to a voice menu application in order to modify it while someone is modifying that application through voice prompt maintenance, the administration terminal will display a message informing you that you can not gain access to the application at this time. Conversely, if someone tries to gain access to an application through the voice prompt maintenance service while it is being modified through the administration terminal, he or she will hear a voice message indicating that the application can not be accessed at this time.

Each voice menu application that you create (announcements, thru-dialers, time-of-day controllers, voice menus) must have a unique ID number which distinguishes from all others. Furthermore, you can assign two different passwords to each application:

- an access password to limit access to menus and prompts: callers who wish to access the application must know this password in order to gain access to the application.
- an update password which, if created, must be known by those who try to access the application through the remote voice prompt maintenance feature.

Announcement Definitions

An announcement is a recording which provides information to callers who dial a predetermined DN. A caller who is connected to an announcement cannot perform any actions, like pressing a key. The caller simply listens to the announcement and hangs up.

Announcements can be played to callers within voice menus or they can be used as a stand-alone voice service (i.e., accessed directly by dialing a DN that is defined in the VSDN table). For example, you may want to record an announcement detailing new health plan benefits for your organization. You can then inform users to dial the DN that will give them access to the recorded information. To use announcements in this manner, as a stand-alone service, a corresponding DN must be defined in the VSDN table.

The Announcement Definitions selection menu (Figure 8-17) displays existing recorded announcements. Announcements are sorted by ID.

If an announcement is set up as a stand-alone voice service (see “The Voice Services-DN Table” earlier in this chapter), the following keys have the functions described below if they are pressed during playback:

1-9	cause the following error message to be played: *“That selection is not recognized.”*
0	calls the Revert DN as defined when you create the announcement
* and #	cause the announcement to be repeated, or if in a voice menu, # causes a return to the voice menu.

Figure 8-17xxx
The Announcement Definitions screen

```
                      VS Config/Menu Applications Admin

 Announcement Definitions

           ID        Title

           211       Printer Status
           212       System Status
           213       Benefits Change
           214       Department Reorganization
           215       Club Activities
           216       Overtime Policy
           217       Health and Safety
           218       Security Code

Move the cursor to the item and press the space bar to select >
    Exit          Add          View/          Delete
                               Modify
```

Procedure 8-15xxx
Using the Announcement Definitions screen

Starting point : The VS Config/Menu Applications Admin Menu, 4 entered.

1 The Announcement Definitions screen appears (Figure 8-17).

2 Choose step 2a to create, 2b to modify, 2c to delete an announcement, or 2d to return.

a. Use [Add].

See the next section, “Adding an Announcement”, for details.

b. Move the cursor to the announcement you require, press <Space Bar> to select, and press [View/Modify].

See “Viewing/Modifying an Announcement” later in this chapter for details.

c. Move the cursor to the announcement you require, press <Space Bar> to select, and press [Delete].

See “Deleting an Announcement” later in this chapter for details.

d. Use [Exit].

The VS Config/Menu Applications Admin menu appears.

Adding an Announcement

Use the Add an Announcement Definition screen (Figure 8-18) to create an announcement and define its general characteristics. An announcement can be made accessible in several ways:

- If the application consists of a single announcement and you want it to be directly dialable, create a voice service DN in the VSDN table and assign it to AS (Announcement Service).
- Include the announcement in a voice menu. Modify the Voice Menu definition so that the announcement ID is associated with a particular key. When a caller presses the key, the announcement is played.
- Include the announcement in a time-of-day controller. Modify the time-of-day controller definition so that the announcement ID is associated with time of day at which it is to be played.

Figure 8-18xxx
The Add an Announcement Definition screen

```
                 VS Config/Menu Applications Admin
 Add an Announcement Definition

 Announcement ID: __________________    Title: _________________

 Revert DN:        ___________________________

 Access Password:    ________________   Update Password:______________

 Announcement  Recorded (Voice):    No

* Language for Prompts:             [AmericanEnglish]
                                    CanadianFrench
                                    AmericanSpanish
                                    French

 Select a softkey >
   Save        Cancel                    Voice
```

* This field appears only if more than one language is installed on the system. The languages displayed here are for illustration purposes only.

The following fields are displayed:

- ***Announcement ID*** - This is a numeric identifier, up to 8 digits in length. It must be unique among all service definitions in order to uniquely identify this announcement.

  You will use this ID when you:

  - need to perform remote maintenance on this announcement;

- want to include this announcement in a voice menu;
- want to use this announcement in a time-of-day controller;
- create a service DN for the announcement in the VSDN table.

- ***Title*** - The name of the announcement, up to 29 characters in length. This title does not need to be unique, but cannot include the characters "+", "?", and "_" because these characters are retrieval-specific characters. It should give you a good indication of the information contained in the announcement.
- ***Revert DN*** - The DN to which the call is routed when the subscriber presses "0" during playback. You may enter up to 30 of the following characters:

0...9	Dialed digits
#	Used by SkyPager
*	3 second pause in dialing
$	Formatting purposes only
( )	Formatting purposes only
_	Formatting purposes only (underscore)
-	Formatting purposes only (hyphen)
spaces	Formatting purposes only

The formatting characters can be used to make the screen display easier to read. (These characters only show up in the screen, they do not generate any corresponding tones.) For example it is easier to decipher 9-(416)-555-1212 than 94165551212. Note that the first character cannot be $.

Note: Common carriers may use some of these formatting characters for other purposes. For example, number sign (#) is used by SkyPager as the numeric pager data terminator character. You may want to contact your common carrier to see if they use any of the other formatting characters.

- ***Access Password*** - This password is optional. If defined, callers need to know this password to hear the announcement. Create a password if you want to create an announcement containing information that only certain people are authorized to hear. The minimum password length is a system-wide parameter. It is defined in the Voice Security Options screen. The maximum password length is 16 characters. Spaces are not allowed. This field is blank by default.

- ***Update Password*** - This field is optional. Define a password if you want to be able to update this announcement from a DTMF telephone set using the Voice Prompt Maintenance service. If the update password is not defined, the Voice Prompt Maintenance service cannot be used to update the announcement. Only the administrator will be able to update the announcement from this form. The minimum password length is a system-wide parameter. It is defined in the Voice Security Options screen. The maximum password length is 16 characters. No spaces are allowed. This field is blank by default.
- ***Announcement Recorded -*** This field indicates whether or not the announcement has been recorded. If it has not yet been recorded, you may do so from this form by using the [Voice] softkey.
- ***Language for Prompts -*** This field is only displayed on those systems that have more than one language installed. The *system* prompts will be played in the selected language. (Note that this is not a translation for your custom prompts and announcements.)

Procedure 8-16xxx
Adding Announcements

Starting point : The Announcement Definitions selection menu, [Add] pressed.

The Add an Announcement Definition screen appears (Figure 8-18).

1 Enter the Announcement ID, the Title, and the Revert DN of the new announcement.

2 Enter an Access Password and an Update Password if required.

3 Use the [Voice] softkey to record the announcement.

See the chapter “Making recordings” for details.

4 Go to 4a to save the new announcement or 4b to cancel the addition.

a. Use [Save].

The system saves the new announcement and redisplays the Announcement Definitions selection menu.

b. Use [Cancel].

The new announcement is discarded and the Announcement Definitions selection menu is redisplayed.

Viewing/Modifying an Announcement

The View/Modify an Announcement Definition screen (Figure 8-19) lets you modify an existing announcement. An administrator and a delegate (using the Voice Prompt Maintenance Service) cannot update the same announcement simultaneously. While an announcement is being updated, the old version of the announcement is still played to callers; the new version takes effect when the updated announcement has been saved. See the section "Adding an Announcement" earlier in this chapter for field descriptions.

Figure 8-19xxx
The View/Modify an Announcement screen

```
                    VS Config/Menu Applications Admin

View/Modify an Announcement Definition

Announcement ID:    4553000_______   Title: Job Listing___________

Revert DN:          2303___________________

Access Password:    0598____________ Update Password:115077_______

Announcement  Recorded (Voice):    No

*Language for prompts:             [English]
                                    French

------------------------------------------------------------------------
   Save        Cancel                      Voice
```

* This field appears only if more than one language is installed on the system. The languages displayed here are for illustration purposes only.

Procedure 8-17xxx
Viewing/Modifying Announcements

Starting point : The Announcement Definitions selection menu, [View/Modify] pressed.

The View/Modify an Announcement Definition screen appears (Figure 8-19).

1 View/Modify the Announcement ID, Title, and Revert DN of the announcement, as required.

2 Change the Access Password and Update Password, if required.

3 Use [Voice] to re-record the announcement.

See the chapter "Making recordings" for more information.

4 Go to 4a to save the announcement or 4b to cancel the modifications.

a. Use [Save].

The system saves the changes and redisplays the Announcement Definitions selection menu.

b. Use [Cancel].

The changes are discarded and the Announcement Definitions selection menu is redisplayed.

Note: Any changes to the voice recordings are also discarded.

Deleting an Announcement

Use the Delete an Announcement Definition form (Figure 8-20) to delete an announcement. If a delegate is modifying the announcement through Voice Prompt Maintenance, you cannot delete the announcement until the delegate is finished. See “Adding an Announcement” earlier in this chapter for field descriptions.

> ***Note:*** If the announcement you are deleting is associated with a voice menu or time-of-day controller, be sure to modify the appropriate definitions to account for the absence of this announcement (i.e., remove references to this announcement ID within voice menu and time-of-day controller definitions). If the announcement is associated with an entry in the VSDN table, you can delete it and then assign the DN to another service.

Figure 8-20xxx
The Delete an Announcement Definition screen

```
                    VS Config/Menu Applications Admin

  Delete an  Announcement Definition

  Announcement ID:    4553000            Title: Job Listing

  Revert DN:          2303

  Access Password:    0598               Update Password:115077

  Announcement  Recorded (Voice):    No

* Language for prompts:              English
                                     Spanish

 OK to       Cancel
 Delete
```

```
* This field appears only if more than one language is on the
  system.
```

Procedure 8-18xxx
Deleting Announcements

Starting point : The Announcement Definitions selection menu, [Delete] pressed.

The Delete an Announcement Definition screen appears (Figure 8-20).

1 Go to 1a to delete the announcement or 1b to cancel the deletion.

a. Use [OK to Delete].

The system deletes the announcement and redisplays the Announcement Definitions selection menu.

b. Use [Cancel].

The announcement is kept and the Announcement Definition selection menu is redisplayed.

Thru-Dialers

A thru-dialer allows callers to dial (make their own call) out of a voice menu application. The type of call they can make is based on how the thru-dial service is defined. Typically, callers are able to dial an extension that is on the switch, or if name dialing is enabled, they can enter the name of the person they want to reach by spelling it out on their touch-tone phone keypad.

Example:

A caller is in a voice menu, listening to a series of choices: *"Press 1 to speak with a sales representative. Press 2 to hear this week's product specials. Press 3 if you know the extension of the person you want to speak with."* In this example, option 3 has been defined as a thru-dialer which provides the caller with access to any extension within the company. When the caller presses 3, he or she is prompted for an extension number. If the caller tries to place a local call (external to the switch), the system will not allow it. The caller hears a message saying *"That number is not authorized. Please try again"*.

In this example, the thru-dialer has been configured in the following way:

- Name dialing is not allowed since the caller is only prompted for an extension. The *Dial By* field in the Add a Thru-Dial Definition screen has been set to Number.
- The restriction codes that have been applied to this thru-dialer do not allow calls external to the switch. Only local extensions are permitted. It is very important to apply restriction codes to thru-dialers in order to avoid toll fraud. You can either apply a predefined set of restriction codes (as configured in the Voice Security Options screen) or you can create a customized set of restriction/permission codes for each thru-dialer that you create (in the Add a Thru-Dial Definition screen).

It is only necessary to define one thru-dialer for each kind of call you want to allow callers to make. For example, you can create thru-dialers for:

- in-house calls only;
- in-house calls and local calls;
- in-house calls and ESN calls;

The "type" of thru-dialer you create is really dependent on the way in which you configure the restriction/permission codes. Because each thru-dialer will have its own unique ID, you can use the same thru-dialer in the various voice menus you create.

It is important that you create a custom greeting for every thru-dialer. This greeting should identify your organization and inform callers what to do. It should also include a message for callers without touch-tone phones. For example, "*If you don't have a touch-tone phone, please stay on the line. An operator will be with you shortly*."

Once a thru-dialer has been created, it can be made accessible in one of three ways.

- Use it as a stand-alone service that is directly dialable. This requires that you create a voice service DN in the VSDN table and assign it to TS (Thru-Dial Service).
- Include it in a voice menu. By associating the thru-dialer with a menu key, callers can dial an extension DN (or enter a name) from within the voice menu. To set this up, you will have to modify the voice menu definition so that the thru-dialer ID is associated with a particular key. When a caller presses the key, he or she is given the chance to specify the extension (or name) of the person they want to talk to.
- Include the thru-dialer in a time-of-day controller. For example, you could create an automated attendant for after-hours by associating the thru-dialer ID with off hours. People making calls after business hours are prompted to enter the extension (or name) of the person they want to reach. Modify the time-of-day controller definition so that the thru-dialer ID is associated with a certain time of day.

Note: If you are using a thru-dialer to accept AMIS networking calls, make sure that the *Short Disconnect* field in the Voice Services Profile is set to at least 10 seconds (10 seconds is the default). This value determines how long the system will wait for a response (telephone keypad entry) before disconnecting a call. Otherwise, an AMIS call that connects to a voice menu or thru-dialer may be prematurely disconnected.

The Thru-Dial Definitions selection menu

In the Thru-Dial Definition screen, thru-dialers are sorted according to service ID.

Figure 8-21
The Thru-Dial Definitions selection menu

```
                VS Config/Menu Applications Admin

 Thru-Dial Definitions


           ID             Title

           3487           Access to ESN
           3488           Access to all numbers
           3490           Access to all numbers
           3510           Automated Attendant
           4356           Access to Long-distance Calls
           6772           Access to ESN
           6778           Access to Local Calls
           8989           Automated Attendant



Move the cursor to the item and press the space bar to select >
    Exit          Add         View/         Delete
                              Modify
```

Procedure 8-19xxx
Using the Thru-Dial Definitions screen

Starting point : The VS Config/Menu Applications Admin Menu, <5> entered.

The Thru-Dial Definitions selection menu appears (Figure 8-21).

1 To create a thru-dialer, choose step 1a. To modify an existing thru-dialer, choose step 1b. To delete a thru-dialer, choose step 1c. To exit the Thru-Dial Definitions selection menu, choose step 1d.

a. Use [Add].

See the next section, “Adding a Thru-Dial Definition”, for details.

b. Move the cursor to the thru-dialer you require, press <Space Bar> to select, and press [View/Modify].

See “Viewing/Modify a Thru-Dial Definition” later in this chapter for details.

c. Move the cursor to the thru-dialer you require, press <Space Bar> to select, and press [Delete].

See “Deleting Thru-Dial Definition” later in this chapter for details.

d. Use [Exit].

The VS Config/Menu Applications Admin menu appears.

Adding a Thru-Dial Definition

The Add Thru-Dial Definition screen (Figure 8-22) is used to create a thru-dialer and define its general characteristics. Once a thru-dialer is created, it should either be associated with a DN in the VSDN table, a voice menu or time-of-day controller.

Figure 8-22xxx
The Add Thru-Dial Definition screen

```
                        VS Config/Menu Applications Admin

 Add a Thru-Dial Definition

   Thru-Dial ID:     _________________   Title: ______________________

   Revert DN:

   Access Password: ___________________ Update Password:  _____________

   Greeting Recorded (Voice):     [No]

 # Language for Prompts:          [AmericanEnglish]
                                  CanadianFrench
                                  AmericanSpanish

 * Dial by:                       Number  Name  [Both]

   DN Length:                     Variable [Fixed] **Digits: ___Left Pad:__

*** Restriction/Permission Set:   [Custom]  OnSwitch  Local  LongDistance1
                                  LongDistance2

   Restriction Codes:  0___ 1___ 2___ 3___ 4___ 5___ 6___ 7___ 8___ 9___

   Permission Codes:   ____ ____ ____ ____ ____ ____ ____ ____ ____ ____

 ________________________________________________________________________
 Select a softkey >
     Save        Cancel                      Voice
```

```
# This field is displayed if multiple languages are installed on the
  system.
* "Name" and "Both" are displayed only if Name Dialing is enabled. This
  would not be applicable if Call Answering is installed.
**"Digits" and "Left Pad" are displayed only if DN length is "Fixed".
***The set names displayed in this field are defaults and may be
  different on your system.
```

The following fields are displayed:

- ***Thru-Dial ID*** - This is a numeric identifier, up to 8 digits in length. It must be unique among all service definitions in order to uniquely identify this thru-dialer.

You can refer to this ID:

- in a voice menu definition - if you associate the thru dial service with a key action, you are prompted for the thru dial ID;
- in a time-of-day controller - this ID can be entered as either the business hours service ID, the off-hours service ID or the holiday service ID;
- in the VSDN table, when you create an access DN so that the thru-dialer can be dialed directly

- ***Title*** - The name of the thru-dialer, up to 29 characters in length. This title does not need to be unique, but cannot include the characters "+", "?", and "_" because these characters are retrieval-specific characters. It should give you a good indication of the function of the thru-dialer.
- ***Revert DN*** - The DN to which the call is routed when the subscriber presses "0" or if a timeout occurs (i.e., the caller stays on the line for a certain period of time without performing any action). You may enter up to 30 of the following characters:

0...9	Dialed digits
#	Used by SkyPager
*	3 second pause in dialing
$	Formatting purposes only
( )	Formatting purposes only
_	Formatting purposes only (underscore)
-	Formatting purposes only (hyphen)
spaces	Formatting purposes only

  The formatting characters can be used to make the screen display easier to read. (These characters only show up in the screen, they do not generate any corresponding tones.) For example it is easier to decipher 9-(416)-555-1212 than 94165551212. Note that the first character cannot be $.

  Note: Common carriers may use some of these formatting characters for other purposes. For example, number sign (#) is used by SkyPager as the numeric pager data terminator character. You may want to contact your common carrier to see if they use any of the other formatting characters.

- ***Access Password*** - This password is optional and can be used to restrict use of the thru-dialer. If defined, subscribers will be required to to enter this password before using this service.

For example, you have created a general-purpose automated attendant using voice menus. One of the menu options is a thru-dialer which allows external callers to dial internal extensions only. However, staff may call in to the automated attendant from outside the office and need to place a long-distance call. To allow for this, you can include a second thru-dial service in the voice menu which allows callers to place long distance calls, but restrict access to this thru-dialer by giving the access password to only those staff who are authorized to place such calls.

The minimum password length is determined by the setting configured in the *Minimum Password Length* field in the Voice Security Options screen. The maximum password length is 16 characters. Spaces are not allowed. This field is blank by default.

- ***Update Password*** - This field is optional. Define a password if you want administrator delegates to be able to update this thru-dialer from a DTMF telephone set using the Voice Prompt Maintenance service. If the update password is not defined, the Voice Prompt Maintenance service cannot be used to update the thru-dialer and only the administrator can update the thru-dialer from this form. The minimum password length is a system-wide parameter. It is defined in the Voice Security Options screen. The maximum password length is 16 characters. No spaces are allowed. This field is blank by default.
- ***Greeting Recorded -*** This field indicates whether or not a custom greeting has been recorded for this thru-dialer. A custom greeting can be recorded from this screen by using the [Voice] softkey or by using the Voice Prompt Maintenance Service. If there is no custom greeting, a default system greeting is used.

  If no custom greeting is recorded, the system greeting that is played depends on the selection made in the *Dial By* field:

  If the *Dial By* field is set to “Number” and if the DN length is fixed, the following system prompt is played: *“Please enter the number of the extension you wish to dial. If you need assistance, just press zero.”*

  If the *Dial By* field is set to “Number” and if the DN length is variable, the following system prompt is played: *“Please enter the number of the extension you wish to dial, followed by number sign. If you need assistance, just press zero.”*

  If the *Dial By* field is set to “Name”, the following system prompt is played: *“Please enter the name of the person you wish to reach, followed by number sign. To enter a name, spell the last name and then spell the first name.”*

If the *Dial By* field is set to “Both”, the following system prompt is played: *“Please enter number or the name of the person you wish to reach, followed by number sign. To enter a name, press 1-1, spell the last name and then spell the first name.”* (Where 1-1 is the name dialing prefix.)

If you record a custom greeting, inform callers whether you want them to dial the extension by name, or number, or if both are acceptable. If callers can dial either the name or number, remember to inform them that they will have to enter the name dialing prefix before spelling out the name on their telephone keypad. This greeting should also contain special information for callers that do not have touch-tone phones. Because rotary phone users can not enter a name or number on a keypad, ask them to wait on the line for operator assistance. You might also inform callers that if they have a touch-tone phone they can press “0” to speak with an operator.

- ***Language for Prompts*** - This field is only displayed if more than one language is installed. If Call Answering (VMUIF) is installed on the system, the selection made here determines the language in which *system* voice prompts are played to callers. If Voice Messaging (MMUI) is installed, it affects the language in which system voice prompts and the default call answering greeting are played to callers. (This does not affect your custom greetings.)
- ***Dial By*** - This field allows you to select the method by which callers dial extension numbers. “Name” and “Both” are displayed only if the Name Dialing feature is installed.

  Note: Name dialing is not available if the VMUIF interface is enabled on the system. Therefore, callers can only dial by number (the “Name” and “Both” options are not offered).

  - ***Number*** - The system interprets digits received from the keypad as numbers. If this method is chosen, the field *DN Length* is displayed (see the description below).
    - ***DN Length*** - This field is displayed only if the previous field, *Dialing By,* is set to “Number”. DNs can either be of variable or fixed length. If all of your DNs are of the same length, use “Fixed”. When “Fixed” is selected, two additional fields, “Digits” and “Left Pad”, are displayed.

      A special situation occurs in hotels that often requires the use of *pad characters*. In a hotel, DNs are based on room numbers. However, room numbers usually vary in length within a hotel. For example, a hotel’s numbering plan goes from Room 101 on the first floor to Room 1999 on the 19th floor.

On the switch, the DNs will be of a fixed length and all three-digit room numbers are preceded by a common number (usually "7"). Thus, the DN length in this hotel would be "Fixed" at "4" digits. The "Left Pad" field represents the common digit that prefixes room numbers that are less than four digits in length. When a guest dials "101" the system uses the pad character to generate "7101" which is the DN that is actually configured on the switch.

When the DN length is fixed, the following fields are displayed:

Digits - The length of the DN. For example, if this field is set to "4", the call is placed right after the fourth digit is entered. This field is not displayed if DN Length is variable.

Left Pad - When the dialed DN is shorter than the fixed DN length, the digit(s) entered in this field are used as a prefix to generate the required DN. The pad character(s) must match the DN configuration on the switch. If a pad character is not required, leave this field blank. This field is not displayed if DN Length is variable.

- ***Name -*** The system interprets digits received from the keypad as letters. The system matches the letters to the subscriber Directory of Last Names. Once the system finds a unique occurrence of the name, DMS VoiceMail will either spell out the name or speak the subscriber's personal verification (if recorded). If more than one person has the same last name, DMS VoiceMail will prompt the caller to enter more of the name if they know it. If DMS VoiceMail finds more than one occurrence of the full name, the caller will be asked to press 1 then 2, and so on, for each occurrence found. This will play the person's personal verification (if recorded). The caller can then choose which person they want to speak to and press the appropriate key to place the call. (This option is not available if the interface is VMUIF.)
- ***Both -*** When this dialing method is selected, a caller can dial through to an extension either by entering an extension number or a name. However, the caller must specify which method they are going to use before entering the name or number. DMS VoiceMail therefore prompts callers to enter a dialing prefix if they are going to dial the name (the default name dialing prefix, 11, unless it has been changed in the Voice Messaging Options form). To dial a number, callers dial the extension followed by "#". (This option is not available if the interface is VMUIF.)

Note: In your custom greeting, be sure to tell callers that if they want to enter a name, they must precede the name with the name dialing prefix (default is "11").

- ***Restriction/Permission Set -*** Choose the restriction/permission set that is to apply to this thru-dialer. You can choose one of the sets that has already been defined or you can create a custom set of codes for this thru-dialer only. (To define codes just for this thru-dialer, select "Custom" and then specify the dialing codes in the fields below.) To see the actual codes defined in the existing restriction/permission sets, review the Voice Security Options screen.

  The restriction/permission set that you choose will determine which dialing codes are permitted and which codes are restricted when a caller dials an extension or external number while connected to this thru-dialer. The set names as shown in Figure 8-22 are the default names and will be different on your system if they have been renamed.

- ***Restriction/Permission Codes -*** These fields are displayed only if "Custom" is selected for *Restriction/Permission Set*. You can define up to 10 restriction and 10 permission codes for each thru-dialer you create.

  The codes that you enter in these fields are access codes. These are the numbers that must be entered in order to dial out of the system to make a local or long-distance call (including area codes). You can also enter extension DNs if you want to prevent callers from dialing certain extensions, such as that of the president or vice-president of your organization.

  Note: By default, all of the restriction fields are prefilled with an integer from 0 to 9 thus restricting all DNs. Therefore, the thru-dialer will be unusable until you have modified these fields.

  For an in depth discussion of restriction and permission codes, see the section "Restriction/permission codes" earlier in this chapter.

Procedure 8-20xxx
Adding Thru-Dialers

Starting point : The Thru-Dial Definitions selection menu, [Add] pressed.

The Add Thru-Dial Definition screen appears (Figure 8-22).

1 Enter the Thru-dial ID, Title, and Revert DN of the new thru-dialer.

2 Set the Access Password and Update Password if required.

3 Set the Language for System Prompts, and Dialing Methods as required. Enter any required Restricted Dialing Prefixes.

4 Use [Voice] to record a customized thru-dialer greeting, if necessary.

See the chapter "Making recordings" for more information.

5 Go to 5a to save the new thru-dialer or 5b to cancel the addition.

a. Use [Save].

The system saves the new thru-dialer and redisplays the Thru-Dial Definitions selection menu.

b. Use [Cancel].

The new thru-dialer is discarded and the Thru-Dial Definitions selection menu is redisplayed.

Viewing/Modifying a Thru-Dial Definition

The View/Modify a Thru-Dial Definition screen (Figure 8-23) lets you modify an existing thru-dialer. An administrator and a delegate cannot update the same thru-dialer greeting simultaneously. While a thru-dialer greeting is being updated, the old version of the greeting is still played to callers. The new version takes effect when the updated greeting has been saved; in the case of Voice Prompt Maintenance, the saving action occurs when the delegate moves on to a new service definition or hangs up.

Figure 8-23xxx
The View/Modify a Thru-Dial Definition screen

```
                    VS Config/Menu Applications Admin

   View/Modify a Thru-Dial Definition

   Thru-Dial ID:     333100________      Title: Local Calls___________

   Revert DN:        2234_____________

   Access Password: ___________________ Update Password:  ____________

   Greeting Recorded (Voice): [No]

 # Language for Prompts:       [AmericanEnglish]
                               CanadianFrench
                               AmericanSpanish

 * Dial by:                    Number  Name  [Both]

   DN Length:                  Variable [Fixed] ** Digits:___Left Pad:___

***  Restriction/Permission Set:[Custom]  OnSwitch  Local  LongDistance1
                               LongDistance2

   Restriction Codes:  613_ 1800 ____ ____ ____ ____ ____ ____ ____ ____

   Permission Codes:   ____ ____ ____ ____ ____ ____ ____ ____ ____ ____

 Select a softkey >
   Save        Cancel                    Voice
```

```
# This field is displayed if multiple languages are installed on the
  system.
* “Name” and “Both” are displayed only if Name Dialing is enabled. This
  would not be applicable if Call Answering is installed.
**“Digits” and “Left Pad” are displayed only if DN length is “Fixed”.
***The set names displayed in this field are defaults and may be
  different on your system.
```

Procedure 8-21xxx
Modifying Thru-Dialers

Starting point : The Thru-Dial Definitions selection menu, [Modify] pressed.

The Modify a Thru-Dial form appears (Figure 8-23).

1 Modify the fields, as required.

2 Use [Voice] to record the thru-dialer greeting, if necessary.

See the chapter “Making recordings” for details.

3 Use 3a to save the thru-dialer or 3b to cancel the modification.

a. Use [Save].

The system saves the thru-dialer and redisplays the Thru-Dial Definitions selection menu.

b. Use [Cancel].

The thru-dialer is not modified and the Thru-Dial Definitions selection menu is redisplayed.

Note: Any changes to voice recordings are also discarded.

Deleting a Thru-Dial Definition

Use the Delete a Thru-Dial Definition form (Figure 8-24) to delete a thru-dialer. If a delegate is modifying the thru-dialer greeting through Voice Prompt Maintenance, you cannot delete the thru-dialer until the delegate is finished. See “Viewing/Modifying a Thru-Dial Definition” for field descriptions.

> ***Note:*** If the thru-dialer you are deleting is associated with a voice menu or time-of-day controller, be sure to modify the appropriate definitions to account for the absence of this thru-dialer (i.e., remove references to this thru-dialer ID within voice menu and time-of-day controller definitions). If the thru-dialer is associated with an entry in the VSDN table, you can delete it and then assign the DN to another service.

Figure 8-24xxx
The Delete a Thru-Dial Definition form

```
                VS Config/Menu Applications Admin

  Delete a Thru-Dial Definition

  Thru-Dial ID:     309                 Title: LongDist

  Revert DN:        0

  Access Password: 7788991              Update Password: 234987

  Greeting Recorded (Voice):     No

# Language for Prompts:          AmericanEnglish
                                 CanadianFrench
                                 AmericanSpanish

* Dial by:                       Number  Name  Both

  DN Length:                     Variable Fixed ** Digits:   Left Pad:

***Restriction/Permission Set:  Custom  OnSwitch  Local  LongDistance1
                                 LongDistance2

  Restriction Codes:     613     1800

  Permission Codes:

 Select a softkey >
   OK to      Cancel
   Delete
```

This field is displayed if multiple languages are installed on the system.
* “Name” and “Both” are displayed only if Name Dialing is enabled. This would not be applicable if Call Answering is installed.
**“Digits” and “Left Pad” are displayed only if DN length is “Fixed”.
***The set names displayed in this field are defaults and may be different on your system.

Procedure 8-22xxx
Deleting Thru-Dialers

Starting point : The Thru-Dial Definitions selection menu, [Delete] pressed.

The Delete a Thru-Dial Definition form appears (Figure 8-24)

1 Go to 1a to delete the thru-dialer or 1b to cancel the deletion.

a. Use [OK to Delete].

The system deletes the thru-dialer and redisplays the Thru-Dial Definitions selection menu.

b. Use [Cancel].

The thru-dialer is kept and the Thru-Dial Definitions selection menu is redisplayed.

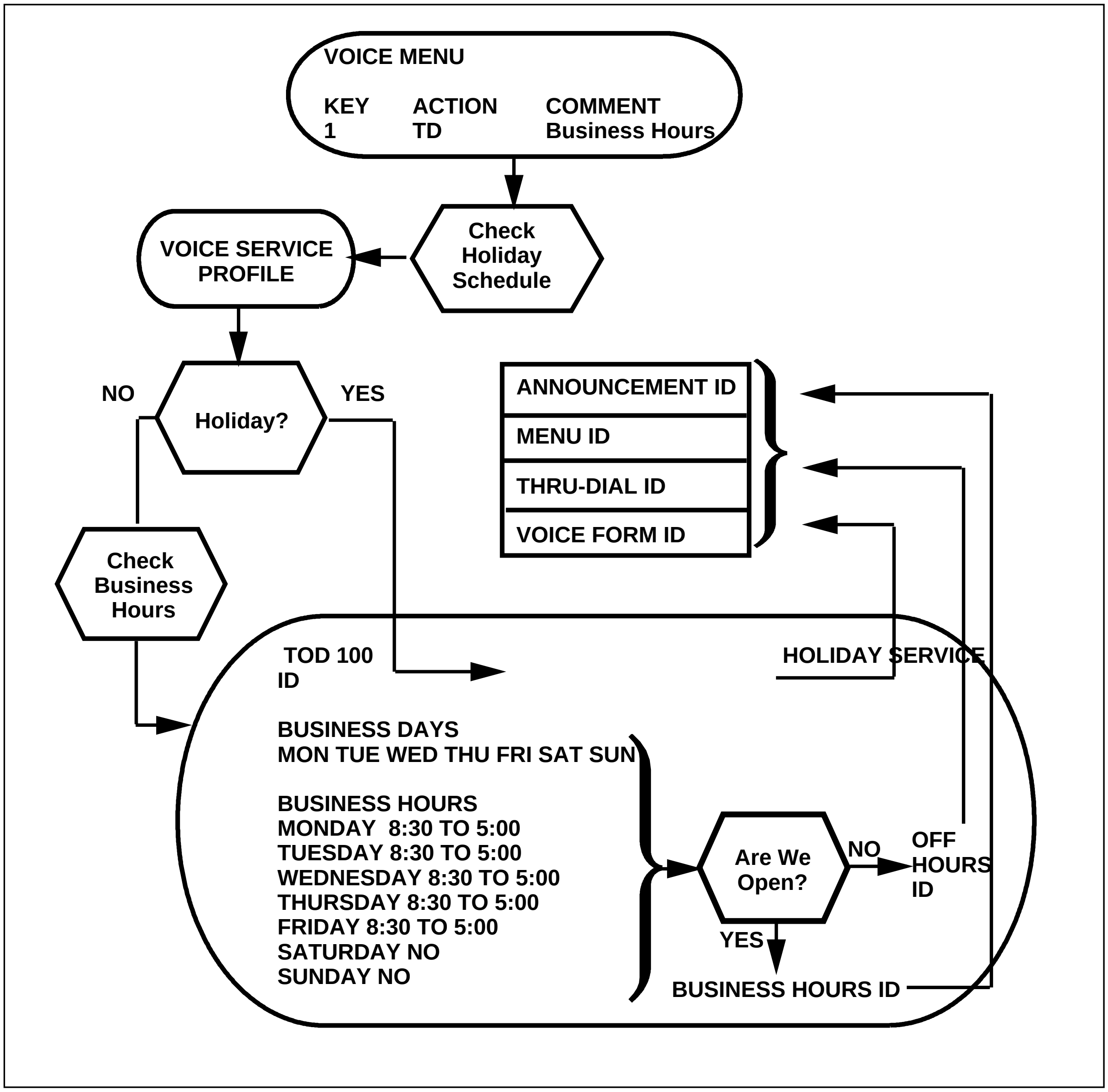
Time-of-Day Controls

A Time-of-Day Control determines the service (menu, announcement, thru-dialer, voice form) that will play based on the day and time of day at which a call is received.

Within the time-of-day control definition, you associate a particular service (a voice menu, announcement, thru-dialer, time-of-day controller, or voice form) with business hours, off hours and holidays. Business days and business hours are defined in the time-of-day control definition. (Default business days and business hours can also be defined in the Voice Services Profile.) Any days which are not defined as business days and times that fall outside of the defined business hours are considered off-hours. Holidays are defined in the Voice Services Profile. A time-of-day controller can be accessed directly or through a voice menu (when a caller presses the key associated with it).

When the time-of-day controller receives a call, it first checks to see if it is a holiday in the Voice Services Profile. If it is a holiday, the holiday service ID is looked up in the time-of-day control definition and the associated announcement, voice menu, thru-dialer or voice form is accessed. If it is not a holiday, the time-of-day controller checks the defined business hours to see if you are open. If you are open, the announcement, voice menu, thru-dialer, or voice form associated with the Business Hours ID is accessed. If you are not open, the service associated with the Off Hours ID is accessed. This is illustrated in Figure 8-25 (in this case the time-of-day controller is accessed via a voice menu when a caller presses “1”).

Figure 8-25xxx
Time-of-day control flowchart



The creation of a time-of-day controller involves the following steps:

1. In the Voice Services Profile, define (a) the default business hours (so that you don't have to define them for every service) and (b) holidays (this includes the start date, end date and start time for each holiday).
2. Create the various announcements, voice menus, thru-dialers, and voice forms (if installed) that will be used.
3. Build the time of day controller. Indicate the service that is to be played during business hours, off-hours and holidays.

4 Activate the time of day controller by associating the TODC ID with a DN in the VSDN table or by including it in a voice menu service (by assigning the TODC ID with one of the menu keys).

5 If the DN you created in step 4 doesn't have a corresponding line DN or UCD queue on the switch, configure one.

6 Test the time of day controller before putting it into service.

CAUTION
Do not reset the system time
To test a time of day controller, do not reset the system time by a few days to "fool" the system into thinking it is the weekend. If you set the time ahead by a number of days, all read messages that meet the Read Message Retention Value (set in the Add Local Voice User screen) will be deleted. For example, today is December 9th and the read message retention limit is 7 days. You set the time ahead by 72 hours. Any messages that are 4, 5 or 6 days old will be deleted before they are supposed to be according to the read message retention maximum.

You can test business hours routing without changing the system time. To test off-hours routing, you can set the time ahead by a few hours (the fewer the better). To test holiday routing, create a holiday with today's date and call the time of day control DN.

Nesting time-of-day controllers

If you have a single facility (for example, a single DID queue) serving multiple departments with different business hours, you can nest several time-of-day controllers in order to meet the needs of all departments. See Appendix B for an example and a flowchart.

The Time-of-Day Control Definitions selection menu

The Time-of-Day Control Definitions selection menu (Figure 8-26) lists all of the time-of-day controllers that have already been built. They are displayed according to Time-of-Day Control service IDs. It is recommended that you keep your own, more detailed list, so that you will know which controller the ID is referring to.

Figure 8-26xxx
The Time-of-Day Control Definitions selection menu

```
                    VS Config/Menu Applications Admin

Time-of-Day Control Definitions


  Time-of-Day       Business Hours    Off-Hours      Holiday
  Control ID        Service ID        Service ID     Service ID

  5550              001               67782          67954
  5551              101               67783          67960
  55632             501               5779           67976
  55636             601               5781           67980

Move the cursor to the item and press the space bar to select >

    Exit          Add          View/          Delete
                               Modify
```

The following fields are displayed:

- ***Time-of-Day Control ID -*** The identification number of the time-of-day controller. This number must be unique across all other voice service IDs.
- ***Business Hours Service ID -*** The identification number of the voice menu application - announcement, voice menu, thru-dialer, time-of-day controller, or voice form (if installed) - that will play during business hours.
- ***Off-Hours Service ID -*** The identification number of the voice menu application - announcement, voice menu, thru-dialer, time-of-day controller, or voice form (if installed) - that will play during off-hours.
- ***Holiday Service ID -*** The identification number of the voice menu application - announcement, voice menu, thru-dialer, time-of-day controller, or voice form (if installed) - that will play during holiday hours.

Procedure 8-23xxx
Using the Time-of-Day Control Definitions selection menu

Starting point The VS Config/Menu Applications Admin Menu, <6> entered.

The Time-of-Day Control Definitions selection menu appears (Figure 8-26).

1 Choose step 1a to create a time-of-day control, 1b to modify an existing Time-of-Day Control, 1c to delete a Time-of-Day Control, or 1d to return.

a. Use [Add].

See the next section, "Adding a Time-of-Day Controller" for details.

b. Move the cursor to the time-of-day control you want to modify, press <Space Bar> to select it, and press [View/Modify].

See "Viewing/Modifying a Time-of-Day Controller" later in this chapter for details.

c. Move the cursor to the time-of-day control you want to delete, press the <Space Bar> to select it, and press [Delete].

See "Deleting a Time-of-Day Controller" later in this chapter for details.

d. Use [Exit].

The VS Config/Menu Applications Admin menu is re-displayed.

Adding a Time-of-Day Controller

The Add a Time-of-Day Control Definition form (Figure 8-27) is used to create a time-of-day controller and define its general characteristics.

Figure 8-27xxx
The Add a Time-of-Day Control Definition screen

```
                    VS Config/Menu Applications Admin
Add a Time-of-Day Control Definition

Time-of-Day Control ID:  _______ Business Hours Service ID:  _______

Off-Hours Service ID:    _______ Holiday Service ID:         _______

       Business Days              Business Hours
       Sunday      [No] Yes
       Monday      No   [Yes]     08:30 to 17:00
       Tuesday     No   [Yes]     08:30 to 17:00
       Wednesday   No   [Yes]     08:30 to 17:00
       Thursday    No   [Yes]     08:30 to 17:00
       Friday      No   [Yes]     10:30 to 13:30
       Saturday    [No] Yes

 Select a softkey >
    Save         Cancel
```

The following fields are displayed:

- ***Time-of-Day Control ID*** - The service ID which uniquely identifies this time-of-day control among all other voice services. The ID can be up to 8 digits in length. No spaces are allowed. You will have to specify this ID when adding a DN to the VSDN table for this time-of-day controller.
- ***Business-Hours Service ID*** - Enter the ID of the voice service - a voice menu, announcement, thru-dialer, time-of-day controller, or voice form (if installed) - that will be provided when a caller dials the DN for this time-of-day control during business hours. The ID can be up to 8 digits in length. No spaces are allowed.

- ***Off-Hours Service ID*** - Enter the ID of the voice service - an existing announcement, voice menu, thru-dialer, time-of-day controller, or voice form (if installed) - that will be provided when a caller dials the DN associated with this time-of-day control during off-hours and non-business days. Note that if a holiday occurs on a non-business day, the Holiday Service ID will be used instead. This ID can be up to 8 digits in length. No spaces are allowed.
- ***Holiday Service ID*** - Enter the ID of the voice menu, announcement, thru-dialer, time-of-day controller, or voice form (if installed) that is to be used when calls are received on a day that has been defined as a holiday to the system. If the ID field is left blank, the system defaults to the Off-Hours Service ID during holiday periods.

  Holidays (up to 20) are defined in the Voice Services Profile (see page 8-59). You can either create an announcement (or other voice menu application) for each holiday or a generic one that covers all holidays. If you create a customized voice menu application for each holiday, remember to change the ID in this field once a holiday has passed so that when the next holiday arrives the appropriate service will be played.

  Note: The specified service is accessed during business hours on the holiday. However, once off-hours begin, the system switches over to the service defined for off-hours. For example, if you create an announcement for Thanksgiving Day, that announcement will be played between 8:30 and 5:00 (the defined business hours). At 5:00 the system presents the service whose ID is defined in the *Off Hours ID* field.

- ***Business Days*** - Define the days of the week that are business days by selecting “Yes”. If a business day turns out to be a holiday, a holiday service will be used instead. The days for which you select “No” will use the service associated with the Off-Hours Service ID, unless overridden by a holiday service.
- ***Business Hours*** - The (default) business hours that you defined in the Voice Services Profile are automatically displayed here. You can override the defaults by entering a new range. The business hours field is not displayed for those days that are defined as non-business days (Saturday and Sunday are defined as non-business days by default). The format is hh:mm to hh:mm using the 24-hour clock. The acceptable range is from 00:00 to 23:59.

Procedure 8-24xxx
Adding a Time-of-Day Control

Starting point : The VS Config/Menu Applications Admin Menu, <6>entered.

The Time-of-Day Control Definitions selection menu appears (Figure 8-26).

1 Press the [Add] softkey.

The Add a Time-of-Day Control Definition screen is displayed.

2 Enter the Time-of-Day Control ID, the Business Hours Service ID, the Off-Hours Service ID and the Holiday Service ID.

3 Modify the defined business days, if required.

4 Modify the defined business hours, if required.

5 To save the new time-of-day controller, go to step 5a. To cancel the changes you have made, go to step 5b.

a. Use [Save].

The system saves the new time-of-day controller and redisplays the Time-of-Day Control Definitions selection menu.

b. Use [Cancel].

The new time-of-day controller is discarded and the Time-of-Day Control Definitions selection menu is redisplayed.

Viewing/Modifying a Time-of-Day Controller

Use the View/Modify Time-of-Day Control definition screen (Figure 8-28) to modify an existing time-of-day controller. The new version takes effect when the updated time-of-day controller has been saved. See "Adding a Time-of-Day Controller" earlier in this chapter for field descriptions.

Figure 8-28xxx
View/Modify Time-of-Day Controller screen

```
                    VS Config/Menu Applications Admin

View/Modify a Time-of-Day Control Definition

Time-of-Day Control ID:      _______   Business Hours Service ID:  ____

Off-Hours Service ID:        _______   Holiday Service ID:         ____

       Business Days                   Business Hours
       Sunday      [No] Yes
       Monday      No   [Yes]          08:30 to 17:00
       Tuesday     No   [Yes]          08:30 to 17:00
       Wednesday   No   [Yes]          08:30 to 17:00
       Thursday    No   [Yes]          08:30 to 17:00
       Friday      No   [Yes]          10:30 to 13:30
       Saturday    [No] Yes

 Select a softkey >

    Save          Cancel
```

Note: The specified service is accessed during business hours on the holiday. However, once off-hours begin, the system switches over to the service defined for off-hours. For example, if you create an announcement for Thanksgiving Day, that announcement will be played between 8:30 and 5:00 (the defined business hours). At 5:00 the system presents the service whose ID is defined in the *Off Hours ID* field.

Procedure 8-25xxx
Modifying Time-of-Day Controllers

Starting point : The VS Config/Menu Applications Admin Menu, <6>selected.

The Time-of-Day Control Definitions selection menu appears (Figure 8-26).

1 Move the cursor to the time-of-day control you want to view or modify.

2 Press <Space Bar> to select it.

3 Press the [View/Modify] softkey.

The View/Modify a Time-of-Day Control Definition screen is displayed.

4 Change any of the following: the Time-of-Day Control ID, the Business Hours Service ID, the Off-Hours Service ID and the Holiday Service ID.

5 Modify the defined business days, if required.

6 Modify the defined business hours, if required.

7 To save the modified time-of-day controller, go to step 7a. To cancel the changes you have made, go to step 7b.

a. Use [Save].

The system saves the modified time-of-day controller and redisplays the Time-of-Day Control Definitions selection menu.

b. Use [Cancel].

The changes you have made to the time-of-day controller are discarded and the Time-of-Day Control Definitions selection menu is redisplayed.

Deleting a Time-of-Day Controller

Use the Delete a Time-of-Day Control Definition form (Figure 8-29) to delete a time-of-day controller. See “Adding a Time-of-Day Controller” earlier in this chapter for field descriptions.

Figure 8-29xxx
The Delete a Time-of-Day Control Definition screen

```
                     VS Config/Menu Applications Admin

Delete a Time-of-Day Control Definition

Time-of-Day Control ID:  45322       Business Hours Service ID: 45999

Off-Hours Service ID:    45988       Holiday Service ID:        45877

       Business Days                 Business Hours
       Sunday      No   Yes
       Monday      No   Yes          08:30 to 17:00
       Tuesday     No   Yes          08:30 to 17:00
       Wednesday   No   Yes          08:30 to 17:00
       Thursday    No   Yes          08:30 to 17:00
       Friday      No   Yes          10:30 to 13:30
       Saturday    No   Yes

 Select a softkey >

  OK to         Cancel
  Delete
```

Procedure 8-26xxx
Modifying Time-of-Day Controllers

Starting point : The VS Config/Menu Applications Admin Menu, <6>selected.

The Time-of-Day Control Definitions selection menu appears (Figure 8-26).

1 Move the cursor to the time-of-day control you want to delete.

2 Press <Space Bar> to select it.

3 Press the [Delete] softkey.

The Delete a Time-of-Day Control Definition screen is displayed.

4 Go to 4a to delete the time-of-day controller or 4b to cancel the deletion.

a. Use [OK to Delete].

The system deletes the time-of-day controller and redisplays the Time-of-Day Controllers selection menu.

b. Use [Cancel].

The time-of-day controller is kept and the Time-of-Day Controllers selection menu is redisplayed.

Voice Menus

The purpose of a voice menu is to assign actions to keys on the telephone keypad. Possible menu actions include playing a recorded announcement, activating a thru-dialer so that the caller can dial an extension by number (or by name), calling a specific number so that the caller can be passed to a person who can answer their question, repeating menu choices, or connecting them to another voice menu or voice service.

A simple voice menu application consists of only one voice menu. More complex applications involve creating several voice menus and linking them together to create several levels of menus.

Once a voice menu has been created, it can be made accessible in one of the following ways:

- Use it as a stand-alone service that is directly dialable. This requires that you create a voice service DN in the VSDN table and assign it to MS (Voice Menu Service).
- Include it in another voice menu to create a multi-level menu application. In this way you can lead users through several layers of menus by associating a menu with a key action in another menu. To set this up, you will have to modify the voice menu definition so that a voice menu ID is associated with a particular key. When a caller presses the key, he or she is connected with the specified menu.
- Include the voice menu in a time-of-day controller. For example, you could create an automated attendant for after-hours that allows callers several choices. To do this, modify the time-of-day controller definition so that the voice menu ID is associated with a certain time of day.

Note: If you are using a voice menu to accept AMIS networking calls, make sure that the *Short Disconnect* field in the Voice Services Profile is set to at least 10 seconds (10 seconds is the default). This value determines how long the system will wait for a response (telephone keypad entry) before disconnecting a call. Otherwise, an AMIS call that connects to a voice menu may be prematurely disconnected.

Recorded prompts

There are three types of recordings that can be made for a voice menu:

- ***A Customized Greeting*** to welcome callers. This is the first prompt that callers hear when they are connected to a voice menu application. (If this is a multi-level menu, this greeting is only played when the caller reaches the first-level menu.) It should (1) identify your organization to let callers know where the call is being answered and (2) inform callers of the menu choices that are currently available.

Always record a greeting for first-level menus. If the voice menu is not a first-level menu, this greeting is not necessary. If no greeting is recorded, the recorded menu choices (see below) are played when a caller accesses the voice menu.

Note: This greeting should also inform callers that they can press star (*) at any time to hear the list of available options.

The following is an example of a greeting:

"Thank you for calling ZUNI Radio. To dial an extension, press 1; to hear the latest weather report, press 2; to talk to the broadcaster, press 3; to leave a song request, press 4. You can hear your current options at any time by pressing star. If you do not have a touch tone phone, wait on the line and an attendant will be with you shortly."

Figure 8-30 shows a flowchart corresponding to this example. Figure 8-32 shows the corresponding Voice Menu Definition.

- ***The Menu Choices Prompt*** explains the options that are available to the caller and solicit an action, after which the system waits for the caller to press a key. If the Greeting and Menu Choices prompt would be identical (for example, if this is not a first-level menu), you only need to record this prompt. These menu choices are played:
  - when a caller accesses a menu that is not a first-level menu;
  - when a caller requests to hear menu choices again (if you have defined the action of one of the menu keys as RP - Repeat Menu Choices);
  - when a caller presses star (*) for assistance;
  - when a caller doesn't enter anything for a specified period of time (that is, it acts as a delayed prompt if RP - Repeat Menu Choices - is defined as the delayed prompt action). Note that a different action can be defined for the initial no response. Initial no response only applies to first-level menus. If a caller doesn't press anything after the greeting in a first-level menu, this is usually a good indication that the caller does not have a DTMF phone. Because the caller cannot press any keys, you might want to revert the caller to a live attendant instead of replaying the menu choices.
  - when a caller accesses a first-level menu that doesn't have a recorded greeting.

  This prompt is not played when a caller first accesses a first-level menu because the greeting includes the menu choices.

  Note: If this is not a first-level menu, this prompt should also inform callers that they can press number sign (#) to return to the previous menu.

The following is an example of recorded menu choices:

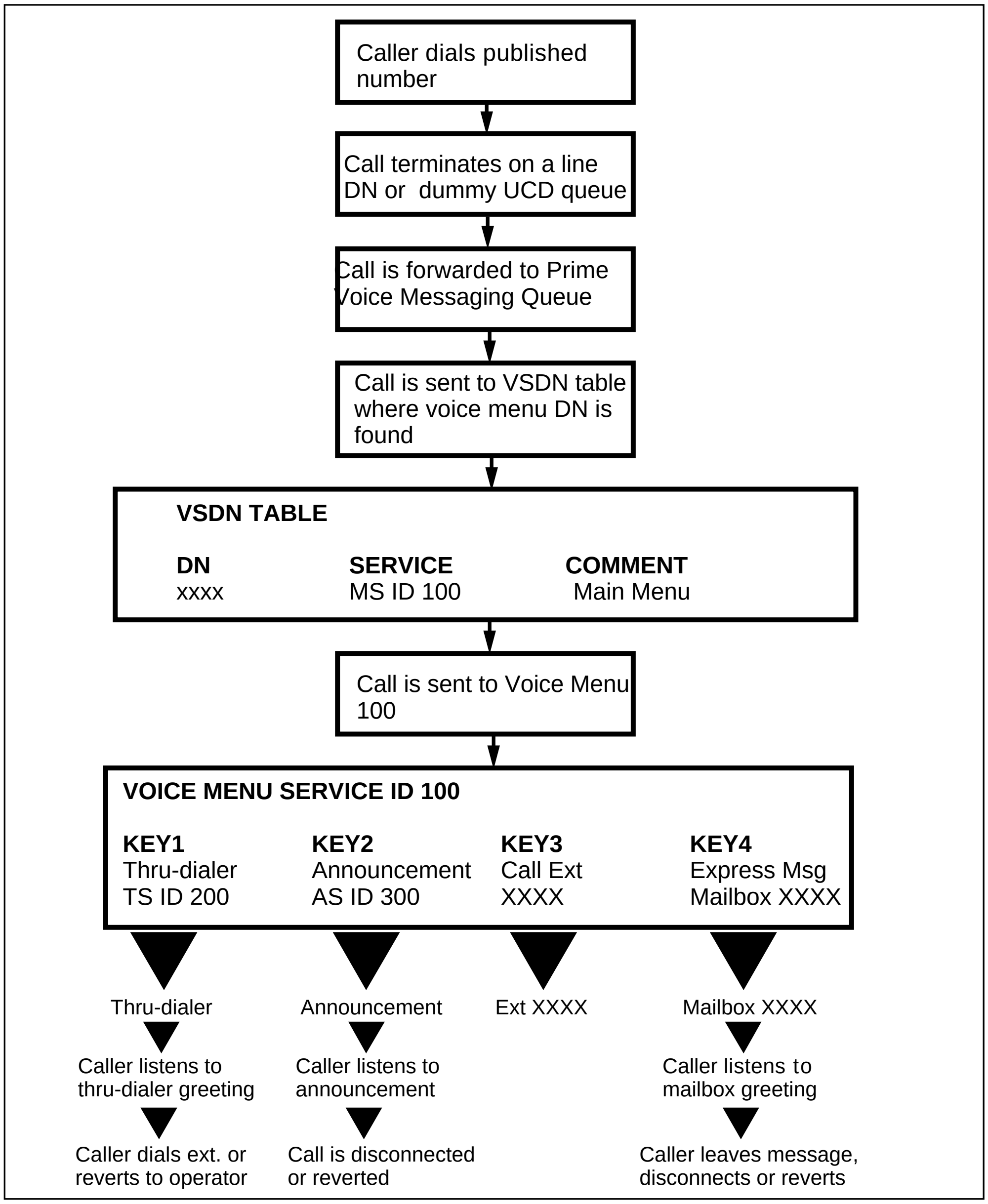
"To dial an extension, press 1; to hear the latest weather report, press 2; to talk to the broadcaster, press 3; to leave a song request, press 4. If you do not have a touch tone phone, wait on the line and an attendant will be with you shortly."

Figure 8-30 shows a flowchart corresponding to this example. Figure 8-32 shows the corresponding Voice Menu Definition.

- ***Play Prompts*** are associated with particular keys on the telephone keypad through the voice menu definition. Play prompts can be used to create customized prompts for any keys in a menu that don't serve any function within the menu. For example, your menu may only have five possible actions (associated with 5 of the keys on the telephone keypad). For the remaining keys you can record a customized prompt that indicates to the caller that the key that was pressed is not valid and request the caller to press a certain key to hear the choices again.

  You can also use play prompts in place of announcements. After a play prompt is played, the caller is still at the same level within the menu and can press one of the other menu options. When an announcement is played, on the other hand, the caller has gone down another level in the menu.

Figure 8-30xxx
Flowchart of a typical voice menu application



A maximum of fourteen actions can be specified for each menu. Three keys produce standard responses (see below) but keys 1 through 9 are administrator-defined. If no menu action has been defined for a particular key, a system error message is played when that key is pressed unless a play prompt has been recorded for that key.

If the caller does not make a selection within a predefined period, one of two default actions is carried out. If the caller has not pressed a key within a set number of seconds after the call has been answered by the menu service (i.e., the caller probably has a rotary phone and cannot press a key), the Initial No Response action is carried out; if the caller has not pressed a key for a defined number of seconds, the Delayed Response action is taken. These actions can be any of the custom menu functions described in the next section, “Adding a Voice Menu”.

Keys that are left undefined in a menu generate a system error message when pressed by the caller. For each key in your menu that is not associated with a service, you should do one of the following:

- Record a customized error message with the Announcement service (AS).
- Record a customized error message with the Play Prompt service (PP).
- Associate the Repeat Menu Choices action (RP) with the key.

Standard menu functions

The digits from 1 to 9 are administrator-definable. Voice menus assign standard functions to the following keys:

0	***Attendant Revert*** - The caller who presses “0” in a voice menu is connected to an attendant. The DN that is called can be specific to each menu (Revert DN).
#	***Return to Previous Menu*** - The caller who presses “#” is returned to the previous menu. If the current voice menu is the first level menu, the first level menu choices are repeated.
*	***Help*** - When a caller presses “*”, the menu choices are played. If no menu prompt exists, the greeting is played.

Custom menu functions

You can assign the following functions to the nine numeric keys on the telephone keypad:

- play a prompt
- play an announcement
- play the menu choices prompt
- go to another menu
- go to the Main Menu
- disconnect the call
- connect the caller to another DN
- request an extension number from the caller
- request a mailbox number (to leave a message)

- play a personal greeting (to leave a message) in a specific mailbox
- prompt the caller to log on to Voice Messaging
- activate a time-of-day controller
- connect to the Revert DN.
- connect to a voice form
- connect to a form transcription service
- connect to a greeting service

  Note: Some of them would depend on certain features being installed, e.g. voice forms.

The Voice Menu Definitions screen

The Voice Menu Definitions selection menu (Figure 8-31) is used to add, modify and delete voice menus. The available voice menus are listed according to their IDs. Each voice menu is assigned a unique identification number.

Figure 8-31xxx
The Voice Menu Definitions selection menu

```
                      VS Config/Menu Application Admin

Voice Menu Definitions


  ID          Title
  7679        Dept. 671
  8081        Managers' Information
  8310        Dept. 603
  9082        Auto Attendant
  9087        Suggested Retail Prices
  9088        Dept. 604
  9164        Salesmen's Schedules
  9166        Customer Information: Line 34

 Move the cursor to the item and press the space bar to select >

    Exit          Add          View/         Delete
                               Modify
```

Procedure 8-27xxx
Using the Voice Menu Definitions selection menu

Starting point : The VS Config/Menu Applications Admin Menu, <7> entered.

The Voice Menu Definitions selection menu appears (Figure 8-31).

1 Choose step 1a to add a voice menu, 1b to modify an existing voice menu, 1c to delete a voice menu, or 1d to return.

a. Use [Add].

See the next section, “Adding a Voice Menu”, for details.

b. Move the cursor to the voice menu you want to view/modify, press <Space Bar> to select it, and press [View/Modify].

See “Viewing/Modifying a Voice Menu” later in this chapter for details.

c. Move the cursor to the voice menu you want to modify, press <Space Bar> to select it, and press [Delete].

See “Deleting a Voice Menu” later in this chapter for details.

d. Use [Exit].

The VS Config/Menu Applications Admin menu is redisplayed.

Adding a Voice Menu

Use the Add a Voice Menu Definition screen (Figure 8-32) to create new voice menus and specify the parameters. Newly created menus take effect once they have been saved.

Figure 8-32xxx
The Add a Voice Menu Definition screen

```
                      VS Config/Menu Applications Admin

  Add a Voice Menu Definition

* Choice of Menu Actions:
  AS Announcement Service    CL   Call                    RV   Call Revert DN
  DS Disconnect              EM   Express Messaging       GS   Greetings Service
  PP Play Prompt             RP   Repeat Menu Choices     MM   Return to Main Menu
  TS Thru-Dial Service       TD   Time-of-Day Control     TR   Transcription Service
  VF Voice Forms Service     MS   Voice Menu Service      VM   Voice Messaging

  Voice Menu ID:    100                    Title:   Main Menu

  Revert DN:        0

  Access Password: 3295341          Update Password: 39243221

  Greeting Recorded (Voice):         [Yes]     Menu Choices Recorded (Voice): [Yes]

  Silent Disconnect:                 [No]    Yes

**Language for Prompts: [AmericanEnglish]
                        CanadianFrench
                        AmericanSpanish
                        French

         Key            Action                               Comments
         1              TS   Thru-Dial ID:      200          Internal thru-dialer
         2              AS   Announcement ID:  300           Weather Report
         3              CL   Calling Number:   3900
                                                             DJ's phone
         4              EM   Mailbox ID:        2339
         5              PP   Recorded Voice:   Yes
         6              PP   Recorded Voice:   Yes
         7              PP   Recorded Voice:   Yes
         8              PP   Recorded Voice:   Yes
         9              PP   Recorded Voice:   Yes
    Initial No Response RP
    Delayed  Response   RV

  Select a softkey >

  Save          Cancel                        Voice
```

* Some of these actions are feature-dependent and may not appear on your screen.
**This field is displayed if multiple languages are installed on your system.

The following fields are displayed:

- ***Choice of Menu Actions*** - This is a read-only list of the actions that can be used in a voice menu. These acronyms are entered in the *Action* field to specify which action to associate with a key on the telephone keypad. For an explanation of these actions, see the description of the *Action* field for details.

  Some of the choices displayed in Figure 8-32 may not appear on your screen. The following features are required to make the corresponding menu actions available:

  - ***V oice Menus*** - **AS** Announcement Service, **TS** Thru-Dial Service, **TD** Time-of-Day Control, **MS** Voice Menu Service
  - ***V oice Forms*** - **TR** Transcription Service, **VF** V oice Forms Service
  - ***Call Answering (VMUIF Interface)*** - **GS** Greetings Service

- ***Voice Menu ID*** - Enter a unique ID, one to eight digits in length (no spaces). This ID must be unique across all other voice services in order to distinguish this voice menu from all other voice menu applications (and voice forms, if installed).

  You can refer to this ID:

  - in another voice menu definition, you can associate this voice menu with a key action in another voice menu in order to create multi-layered menus;
  - in a time-of-day control definition, as either the business hours service ID, the off-hours service ID, or the holiday service ID;
  - in the VSDN table, when you create an access DN to make this voice menu directly dialable (for example, if this is a first-level menu).

- ***Title*** - The name of the voice menu. Do not use the following characters: “?”, “+”, or “_”.

- ***Revert DN*** - The DN to which the call is routed when the subscriber presses “0” or when a timeout occurs. You may enter up to 30 of the following characters:

0...9	Dialed digits
#	Used by SkyPager
*	3-second pause in dialing
$	Formatting purposes only
( )	Formatting purposes only
_	Formatting purposes only (underscore)
-	Formatting purposes only (hyphen)
spaces	Formatting purposes only

The formatting characters can be used to make the screen display easier to read. (These characters only show up in the screen, they do not generate any corresponding tones.) For example it is easier to decipher 9-(416)-555-1212 than 94165551212. Note that the first character cannot be $.

Note: Common carriers may use some of these formatting characters for other purposes. For example, number sign (#) is used by SkyPager as the numeric pager data terminator character. You may want to contact your common carrier to see if they use any of the other formatting characters.

- ***Access Password*** - This password is optional and can be used to restrict use of this voice menu. If defined, users will be required to enter this password before using this service. The minimum password length is defined in the Voice Security Options screen. The maximum password length is 16 characters. Spaces are not allowed. This field is blank by default.
- ***Update Password*** - This field is optional. Define a password if you want administrative delegates to update this voice menu from a DTMF telephone set using the Voice Prompt Maintenance service. If the update password is not defined, the Voice Prompt Maintenance service cannot be used to update the voice menu and only you will be able to update the service from this screen.

  The minimum password length is defined in the Voice Security Options screen. The maximum password length is 16 characters. Spaces are not allowed. This field is blank by default.
- ***Greeting Recorded (Voice***) - When you record a greeting for the menu this field will be set to “Yes”. The default is “No”.
- ***Menu Choices Recorded (Voice) -*** This is the recording in which you specific the relevant keys and the associated actions. For example, *“Press 1 for customer service. Press 2 to speak with a sales representative. Press 3 if you know the extension of the person you wish to speak with.”* When you record the prompt for the menu choices, this field is automatically set to “Yes”.
- ***Silent Disconnect -*** When this field is set to “Y es”, the “Goodbye” prompt is not played when a caller is disconnected from the voice menu. This allows voice menus to be integrated into other applications. Upon disconnection from a voice menu a caller may therefore be passed to a another queue for holding or another voice application. In this case, the voice menu is not the terminating application and if the caller hears the “Goodbye” prompt in this situation, he or she may hang up prematurely. If the voice menu is the terminating application, set this field to “No” to play the “Goodbye” prompt when the caller disconnects. The default is “No”.

- ***Language for Prompts*** - This field is only displayed for those systems that have more than one language installed. System prompts will be played in the selected language.
- ***Key*** - This is a read-only field that displays the keys on the telephone keypad. There are two extra entries, one for Initial No Response and one for Delayed Response. You should assign actions to these two timeout conditions in the *Action* field. (The actual timeout values are configured in the Voice Services Profile field in the *Command Entry* and *Short Disconnect* fields.) The action associated with the Initial No Response timeout applies when a caller has entered a voice menu and has not yet pressed any keys (probably because they have a rotary phone). The action associated with the Delayed Response timeout applies when a caller has indicated that they have a touch-tone phone (they have pressed at least one key while in the menu) but has not pressed a key for a specified number of seconds.
- ***Action*** - Enter the action that is to occur when a caller presses the associated key.

  AS ***Announcement Service*** - When you specify AS, the *Announcement ID* field is displayed. Enter the ID of the announcement you want callers to hear when they press the key you are defining.

  If any keys in this menu are undefined, you can use the announcement service to record a customized error greeting (otherwise a system error message is played). The announcement should inform callers that the key they have pressed is invalid and then inform them of the available menu choices. You will first have to define and record an announcement to be invoked; see “Announcements” earlier in this chapter for details. Alternatively, you can use the Play Prompt service (PP) to record a greeting, or use the RP service to repeat the list of allowed choices.

  Announcements are more ideally suited to inform callers on some specific item other than how to use a menu. In fact, to return from an announcement to the menu, a caller cannot just press any menu key, but must press # while the announcement is playing.

  CL ***Call*** - This action places a call. When you specify CL, the *Calling Number* field is displayed. Enter the number that needs to be dialed when a caller presses the associated key.

  RV ***Call Revert DN*** - When this action is specified, pressing the associated key places a call to the Revert DN defined earlier in this form.

  DS ***Disconnect*** - When the associated key is pressed, the caller is disconnected from the current voice menu application. If Silent

Disconnect is set to “No”, the caller hears a system prompt (“*Goodbye*”) before being disconnected. If Silent Disconnect is set to “Yes”, the caller is disconnected from the current voice menu without hearing the “Goodbye” prompt. This is desired when calls will be connected to another service upon disconnection.

EM ***Express Messaging*** - When the associated key is pressed, express messaging is initiated. If you specify a Mailbox ID, the caller hears the greeting for that mailbox and is prompted to leave a message at the record tone. If you do not specify a Mailbox ID, the caller is prompted for a mailbox number. The greeting for that mailbox (if there is one) is played and the subscriber is prompted to leave a message. The caller cannot return to the voice menu from express messaging and must hang up. This field can hold up to 18 digits to accommodate sites in Meridian Networking systems (CPE only).

GS ***Greetings Service -*** Pressing the associated key connects callers to the service which allows subscribers to update their greetings using a simplified interface that requires no keypad input. The service simply prompts callers to speak at certain times to record a new greeting.

Note: You should create a DN for this service in the VSDN table to service subscribers that have rotary phones (since only those subscribers with digitone phones can access voice menus).

PP ***Play Prompt*** - This action can be selected for any key that does not serve a function in the menu (i.e., it isn’t associated with a service). Play Prompt allows you to record a customized error message for each unused key from this form. The prompt should inform the caller that the key they just pressed cannot be used. It should also give them instructions about how to hear the menu choices again.

When you enter PP in the *Action* field, a new field, *Recorded (Voice)*, is displayed. Initially, it will indicate that there is no recorded prompt. You can use the [Voice] softkey to record and review the prompt. See the chapter “Making recordings” for details. If a custom prompt is not recorded, the caller hears a system error message and is given the chance to make another choice.

Alternatively, you can use the announcement service (AS) to record customized error messages or the RP service to repeat the allowed menu choices.

RP ***Repeat Menu Choices*** - When callers press the associated key, the menu choices prompt is played. This action should be available at every menu level and can also be used for keys that are not associated with any service.

MM ***Return to Main Menu*** - This action takes the caller to the main menu and plays the associated menu choices prompt. If the current menu is the main menu (i.e., the first-level menu), the menu choices prompt is repeated for the main menu.

TS ***Thru-Dial Service*** - This action connects the caller to the thru-dialer specified in the accompanying *Thru-Dial ID* field. The caller can return to the voice menu from the thru-dial service by pressing the number sign (#) on the telephone keypad prior to the call being connected.

TD ***Time-of-Day Control*** - This action activates a time-of-day control that determines which services are available on certain days and at different times of the day. Enter the ID of the time-of-day control in the associated TODC (ToD Control) ID field.

TR ***Transcription Service*** **-** This action connects the caller to the Transcription service in order to collect information collected by a voice form. You are prompted for a voice form ID: this is optional. Enter an ID if you want the caller (a transcriber) to be logged into a particular voice form. Leave the ID field blank if you want to the transcriber to specify the voice form ID.

VF ***Voice Forms Service*** **-** This action connects the caller to a voice form in which they are prompted to answer questions by pressing keys on the telephone keypad. Enter the Voice Form ID in the corresponding field.

MS ***Voice Menu Service*** - This action takes the caller to the voice menu specified in the accompanying *Voice Menu ID* field. If the menu does not exist, an error message is played and the caller can make another choice.

VM ***Voice Messaging*** - This action initiates V oice Messaging and prompts the caller for a Mailbox ID. The caller cannot return to the menu once voice messaging has been invoked.

- ***ID*** - Depending on the action you specified, an additional field may appear in which you will have to enter a service ID, mailbox ID or calling number as explained in the above field descriptions.
- ***Comments*** - You may enter a descriptive comment for each key-action combination.

Procedure 8-28xxx
Adding Voice Menus

Starting point : The Voice Menu Definitions selection menu, [Add] pressed.

The Add a Voice Menu Definition screen appears (Figure 8-32).

1 Enter the Voice Menu ID, Title, and Revert DN.

2 Enter an Access Password and an Update Password, if required.

3 Move the cursor to the *Greeting Recorded* field, and use the [Voice] softkey to record the greeting.

See the chapter "Making recordings" for information about using the [Voice] softkey.

4 Move the cursor to the *Menu Choices Recorded* field, and use the [Voice] softkey to record the menu choices prompt.

See the chapter "Making recordings" for information about using the [Voice] softkey.

5 Enable *Silent Disconnect*, if necessary.

6 If you have a multi-lingual system, select the language in which system prompts are to be played.

7 Enter an Action for each required key; enter any required ID as described above. Enter a Comment describing the action.

8 Go to 8a to save the new voice menu or 8b to cancel.

a. Use [Save].

The system saves the new voice menu and redisplays the Voice Menu Definitions selection menu.

b. Use [Cancel].

The new voice menu is discarded and the Voice Menu Definitions selection menu is redisplayed.

Note: Any new voice recordings are also discarded.

Personal voice menus

Voice menus can be created specially for particular users who require more options than the standard call answering options that are presented to callers (to leave a message or press “0” for assistance). These are known as *personal voice menus*. Discuss with your user the various actions that are required and determine an appropriate design.

The user’s DN is entered as a voice service DN in the VSDN table. When the DN is called, the voice menu application associated with the DN takes the call. Unlike regular voice menus, personal voice menus do not require a line DN on the DMS/SL-100 because there is a real DN associated with the user’s terminal. To set up a personal voice menu, follow the procedure below.

Procedure 8-29xxx
Creating a personal voice menu

1 Create a voice menu application for the user as described in the previous section “Adding a Voice Menu”.

2 Access the Voice Services-DN Table.

3 Press the [Add] softkey.

The Add DN Information screen is displayed.

4 Enter the user’s extension DN as the Access DN.

5 Enter MS (voice menu service) as the Service.

6 Save the DN information.

Viewing/Modifying a Voice Menu

Use the View/Modify a Voice Menu Definition screen (Figure 8-33) to alter an existing voice menu. An administrator and a delegate cannot update the prompts for the same voice menu simultaneously. While a voice menu is being updated using the Voice Prompt Maintenance Service, the old version of the menu is still played to callers. The new version takes effect when the updated voice menu has been saved. See “Adding a Voice Menu” earlier in this chapter for field descriptions.

Figure 8-33xxx
The View/Modify a Voice Menu Definition screen

```
                    VS Config/Menu Applications Admin

  View/Modify a Voice Menu Definition

* Choice of Menu Actions:
  AS Announcement Service    CL   Call                    RV   Call Revert DN
  DS Disconnect              EM   Express Messaging       GS   Greetings Service
  PP Play Prompt             RP   Repeat Menu Choices     MM   Return to Main Menu
  TS Thru-Dial Service       TD   Time-of-Day Control     TR   Transcription Service
  VF Voice Forms Service     MS   Voice Menu Service      VM   Voice Messaging

  Voice Menu ID:    100                      Title:   Main Menu

  Revert DN:        0

  Access Password: 3295341          Update Password: 39243221

  Greeting Recorded (Voice):         [Yes]     Menu Choices Recorded (Voice): [Yes]

  Silent Disconnect:                 [No]   Yes

**Language for Prompts: [AmericanEnglish]
                         CanadianFrench
                         AmericanSpanish
                         French

         Key            Action                            Comments
         1              TS   Thru-Dial ID:     200        Internal thru-dialer
         2              AS   Announcement ID: 300         Weather Report
         3              CL   Calling Number:  3900
                                                          DJ's phone
         4              EM   Mailbox ID:       2339
         5              PP   Recorded (Voice):Yes
         6              PP   Recorded (Voice):Yes
         7              PP   Recorded (Voice):Yes
         8              PP   Recorded (Voice):Yes
         9              PP   Recorded (Voice):Yes
     Initial No Response RP
    Delayed  Response   RV

 Select a softkey >

    Save          Cancel                          Voice
```

* Some of these actions are feature-dependent and may not appear on your screen.
**This field is displayed if multiple languages are installed on your system.

Procedure 8-30xxx
Viewing/Modifying Voice Menus

Starting point : The Voice Menu Definitions selection menu, [View/Modify] pressed.

1 Modify the fields as required.

If some of Key fields cannot be seen, use the Next Scrn hardkey to scroll the Key and Action fields (the Choice of Menu Actions will remain stationary). Use the Prev Scrn hardkey to scroll back up.

2 Move the cursor to the *Greeting Recorded* field, and use the [Voice] softkey to change the greeting if necessary.

See the chapter "Making recordings" for details.

3 Move the cursor to the *Menu Choices Recorded* field, and use the [Voice] softkey to change the menu choices prompt.

See the chapter "Making recordings" for details.

4 Go to 4a to save the voice menu or 4b to cancel.

a. Use [Save].

The system saves the voice menu and redisplays the Voice Menu Definitions selection menu.

b. Use [Cancel].

The modifications to the voice menu are discarded and the Voice Menu Definitions selection menu is redisplayed.

Note: Any changes to voice recordings are also discarded.

Deleting a Voice Menu

Use the Delete a Voice Menu Definition selection menu (Figure 8-34) to delete a voice menu. The fields in this form are read-only.

> ***Note:*** If the voice menu you are deleting is associated with another voice menu or time-of-day controller, be sure to modify the appropriate definitions to account for the absence of this voice menu (i.e., remove references to this voice menu ID within voice menu and time-of-day controller definitions). If the voice menu is associated with an entry in the VSDN table, you can delete it and then assign the DN to another service.

Figure 8-34xxx
The Delete a Voice Menu Definition screen

```
                     VS Config/Menu Applications Admin

Delete a Voice Menu Definition

*Choice of Menu Actions:
AS Announcement Service    CL   Call                  RV   Call Revert DN
DS Disconnect              EM   Express Messaging     GS   Greetings Service
PP Play Prompt             RP   Repeat Menu Choices   MM   Return to Main Menu
TS Thru-Dial Service       TD   Time-of-Day Control   TR   Transcription Service
VF Voice Forms Service     MS   Voice Menu Service    VM   Voice Messaging

Voice Menu ID: 101                     Title: Night Attendant

Revert DN:     0

Access Password: 3341341   Update Password: 9807326

Greeting Recorded (Voice):    No      Menu Choices Recorded (Voice):  No

Silent Disconnect:            No      Yes

**Language for Prompts:  AmericanEnglish
                         CanadianFrench
                         AmericanSpanish
                         French

       Key               Action                           Comments
       1                AS   Announcement ID  2433
       2                 CL   Calling Number:  3900

       3                 DS
       4                 EM   Mailbox ID:      2339
       5                 GS
       6
       7                 MS   Voice Menu ID:   2900
       8                 PP   Recorded (Voice):No
       9                 PP   Recorded (Voice):No
  Initial No Response    RP
  Delayed  Response      RV
________________________________________________________________________________
Select a softkey >

 OK to            Cancel
 Delete
```

* Some of these Actions are feature-dependent and may not appear on your screen.
**This field only shows up if you have multiple languages installed on your system.

Procedure 8-31xxx
Deleting Voice Menus

Starting point : The Voice Menu Definitions selection menu, [Delete] pressed.

The Delete a Voice Menu Definitions screen appears (Figure 8-34).

1 Go to 1a to delete the voice menu or 1b to cancel the deletion.

a. Use [OK to Delete].

The system deletes the voice menu and redisplays the Voice Menu Definitions selection menu.

b. Use [Cancel].

The voice menu is kept and the Voice Menu Definitions selection menu is redisplayed.

Using voice prompt maintenance

The Voice Prompt Maintenance service allows you to re-record prompts used in voice menu applications from a remote touch-tone telephone. This means that you do not have to be at the administration terminal to update recordings. This is especially useful for recordings that are updated frequently.

Note: This service cannot be used to update voice form prompts.

You cannot delete recordings through this service, however, a newly recorded prompt overwrites any previous prompt. While you are in the process of updating a recording, any callers who reach the voice menu application continue to hear the old recording. Callers who reach the application once the new prompt has been saved hear the new prompt.

To make the voice prompt maintenance service available, you will have to define a DN for it in the VSDN table (see Procedure 8-32). Furthermore, this service can only be used to change announcements, voice menu prompts or thru-dialer greetings if an Update Password has been defined for the application. If no password is defined, prompts can only be updated from the administration terminal.

Under certain circumstances, you will have to decide whether to use the voice prompt maintenance service or the remote activation service (described in the next section). For example, the office is unexpectedly closed due to inclement weather and you want to ensure that callers are aware that this is the reason for the office closure. During normal business hours you use an auto-attendant which is a thru-dialer that allows callers to call through to the desired extension. You could either (a) change the thru-dialer greeting with the voice prompt maintenance service or (b) use remote activation to change the service that is accessed when the number is dialed. Of course, you will have to have some sort of announcement already recorded (see the following section). If you choose to change the thru-dialer greeting, you will have to remember to change it back to the original greeting once the office is open. However, if an operator answers the phone during normal business hours, you will probably use the remote activation feature to ensure that an announcement is played.

Procedure 8-32xxx
Defining the voice prompt maintenance DN

Starting point The Main Menu.

1 Select Voice Administration.

2 Select Voice System Configuration/Voice Menu Applications Administration.

3 Select the Voice Services-DN.

4 Press the [Add] softkey. (To modify an existing DN, press [View/Modify].

The Add DN (or View/Modify DN) Information screen is displayed.

5 Enter an Access DN.

6 Enter PM (Prompt Maintenance) in the *Service* field.

7 Enter a comment (if desired).

8 Use step 8a to save the voice prompt maintenance DN definition; go to step 8b to cancel.

a. Use [Save].

The service is defined in the VSDN table.

b. Use [Cancel].

The voice prompt maintenance service remains undefined or unchanged.

Once the voice prompt service has been configured, follow Procedure 8-33 to use the service to update voice menu prompts and Procedure 8-34 to update announcements and thru-dialer prompts.

Procedure 8-33xxx
Updating voice menu prompts

1 Dial the Voice Prompt Maintenance Service DN.

The system prompts you for an ID.

2 Enter the ID of the voice menu you want to modify and press **#**.

The system prompts you for the Update Password.

3 Enter the Update Password and press **#**.

4 The system plays a menu with four choices:

a. Update Greeting prompt

b. Update Menu Choices prompt

c. Update No Response prompt

d. Update Other Menu prompts

5 Select the required function.

If you select a, b, or c you are prompted to play the existing prompt or record a prompt.

Option d allows you to update prompts associated with particular keys. You are prompted to enter the prompt number. Enter the keypad number that a caller must press to hear the prompt.

6 Play or record the prompt.

If you selected d after playing, recording, or updating the prompt, enter a number sign (#) to go back to where you can enter the (key) number of another prompt.

7 To return to the ID prompt, enter a number sign.

You can now work on another menu by going to step 2.

Procedure 8-34xxx
Updating announcements and thru-dialer greetings

The system prompts you for an ID.

1 Enter the required Announcement ID or Thru-dialer ID and press **#**.

The system prompts you for the Update Password.

2 Enter the Update Password and press **#**.

You are prompted to use Play or Record (Use Play to hear the entire prompt from start to finish).

3 Play the announcement or greeting, or update it and save the new announcement.

Record overwrites the old recording.

4 To return to the ID prompt, enter a number sign.

You can update another announcement or thru-dialer greeting by going to step 2.

Using remote activation

The Remote Activation service provides the administrator with the capability to switch voice menu applications (voice menus, announcements, thru-dialers, time-of-day controllers, and voice forms, if installed), through a standard DTMF telephone set. For example, in the case of a storm, an administrator who is off-site may switch an automated attendant menu to a pre-defined announcement informing callers that the office is closed due to the storm. Voice Menu Applications are described later in this chapter.

For example, you may have a regular “closed” greeting that goes as follows:

> *“Thank you for calling the First Bank of Moosejaw. We are open Monday to Friday from 8:00 a.m. to 4:00 p.m. Please call again when we are open.”.*

You create the following “snow storm” greeting in preparation for the upcoming winter:

> *“Thank you for calling the First Bank of Moosejaw. Due to the recent snow storm, our offices will be closed today. We are normally open Monday to Friday from 8:00 a.m. to 4:00 p.m. Please call again when we are open. Thank you.”*

To use the Remote Activation feature you must first define a DN for the Remote Activation service so that an entry is made in the Voice Service-DN Table. This makes the service available through a dialable DN. A password is also required to use the Remote Activation service. This password is defined in the Add DN Information form. Callers dialing the Remote Activation DN cannot access this service without the password.

Note: If the password field is left blank, this effectively disables the remote activation service itself. Therefore, to temporarily disable remote activation, you do not have to delete the RA service from the VSDN table.

You should also create a complete listing of all voice menu services. Keep a copy of the list at home (or wherever you will be calling from to make changes). Make sure the place you choose to keep it is secure. This listing should include the voice service DN, the title (or a brief description of the service) so that you can easily identify it, the update password (if defined), and the access password (if defined).

Remote Activation only allows you to change which service is accessed when a particular DN is dialed. You cannot change any recorded prompts using this service. To change prompts remotely, use the Voice Prompt Maintenance feature.

Procedure 8-35xxx
Defining/Modifying the Remote Activation Service and Password

Starting point : The Voice Administration menu, the Voice System Configuration item selected.

1 Select the Voice Services-DN item from the menu.

2 Press the [Add] softkey.

The Add DN Information form is displayed.

3 Enter an Access DN.

4 Enter RA in the *Service* field.

A Password field is displayed.

5 Enter the password, up to 16 numeric characters in length, that users will require to use the Remote Activation service.

6 Enter a comment (if desired).

7 Use step 7a to save the Remote Activation DN definition; go to step 7b to cancel.

a. Use [Save].

The service is defined in the VSDN table.

b. Use [Cancel].

The Remote Activation Service is not saved.

Once you have defined a DN and a password for Remote Activation you may use the feature to change voice services from a remote DTMF telephone.

Procedure 8-36xxx
Using Remote Activation

Starting point : A DTMF telephone set.

1 Call the Remote Access DN as defined in the Voice Service-DN Table.

You are is prompted for a password.

2 Enter the Remote Activation Password, followed by **#**.

You are prompted to enter a voice service DN.

3 Enter the DN of the voice service you want to change, followed by **#**.

A voice prompt confirms your selection by stating the DN and the associated service ID.

You are prompted to enter a new service ID.

4 Enter the ID of the application you want to associate with the DN entered in step 3.

You will hear a confirmation that the new service is now associated with the DN.

You are then prompted to enter a new service ID. This gives you the opportunity to change the service ID you just entered in case of an error.

5 If you want to change the service ID you entered in step 4, enter the new service ID followed by #.

If you want to change another voice service, enter # and you will be prompted for another voice service DN. Repeat from step 3 for each voice service you want to change.
If no further changes are required, hang up the phone.

When you check the Voice Service-DN Table, you will notice that in the *Comments* field for those services that were changed, a message, “Changed by R.A.”, will appear. The *Service* field will display the service that was newly associated with the DN during the Remote Activation session.

Outcalling Administration

The Outcalling feature provides two services. Remote Notification (RN) informs subscribers via a pager, paging service, or remote telephone that there are new messages in their mailbox. Delivery to Non-user allows an MMUI DMS VoiceMail subscriber to compose and send messages to non-users of DMS VoiceMail.

> ***Note:*** Delivery to Non-users is not available if Call Answering (VMUIF) is installed on the system.

Outcalling administration is performed in two places: in User Administration (when you add or modify a user mailbox) and in Voice Administration, from the Outcalling Administration menu.

From the Outcalling Administration menu, you can choose to:

1 configure outcalling parameters on a system-wide basis in the Outcalling Options screen, or

2 display the Outcalling Audit Trail Report.

Outcalling options

Outcalling administration is done in two places:

1 The Outcalling Options screen.

2 In User Administration, in the Add a Local Voice User screen (Outcalling Fields).

Outcalling options

Some of parameters in the Outcalling Options screen allow you to control how outcalling audit trail data is collected. For example, you can enable or disable the collection of audit trail data, specify how long audit trail data should be stored on disk before being deleted, and the maximum number of channels that can be simultaneously used by outcalling features. Most of the remaining fields are for configuring the Delivery to Non-Users feature. There are also several fields related to Remote Notification, such as the maximum number of retry repeats, the numeric pager data terminator and the default numeric pager data. The parameters configured here apply to all users on the system.

User administration

When you add new users to the system, there are a number of outcalling fields that allow you customize the outcalling and delivery non-user features for each user. This is where you enable or disable outcalling features for a particular user. Most the parameters relate to the remote notification feature. It is in the Add a Local Voice User screen that you create remote notification schedules for users. These schedules determine where the remote notification is sent (for example, to a phone or a pager), the phone or pager number. See the section “Defining Outcalling parameters” in the “User

Administration" chapter. This section also describes how to set up remote notification for the different types of pagers that are supported.

Many of the parameters in the Outcalling Options screen are not displayed if Call Answering is installed on the system. See Figure 8-36.

Table 8-6 divides outcalling administration into specific parameters and user-specific parameters. The user-specific parameters are configured when you are adding new users or modifying existing users in User Administration. If a field is configurable on both a system and user basis (such as DNU DTMF Confirmation Required), the user setting usually overrides the system setting. The specific parameters are accessed through the Outcalling Options screen. See the section "Defining Outcalling parameters" in the "User Administration" chapter.

Table 8-6xxx
System-wide and User-specific Outcalling parameters

Service	System-wide Parameters	User-specific Parameters
RN	•Maximum Number of Remote Notification Retry Repeats •Default Numeric Pager Data (this can be defined differently for each subscriber by filling in the *Numeric Pager Data* field when setting up schedules)	•Remote Notification Capability •Current State (on or off) •Keypad Interface (when enabled, allows subscribers to set up schedules) •Message Notification Options (subscriber notified of Any or Urgent messages) •Restriction/Permission Codes •Retry Limits and Intervals Remote Notification Schedules (Business Days, Non-Business Days, Temporary)
DNU	•Delivery to Non-user Weekdays •Delivery to Non-user Weekends •Delivery to Non-user Retries •Delivery to Non-user Addressing Prefixes and Associated Dialing Codes •Number of Times to Play a Non-user Message •DNU DTMF Override •DNU DTMF Confirmation Required	•Delivery to Non-user Capability •DNU DTMF Confirmation Required •Delivery to Non-user Restriction/Permission Codes

The Outcalling Administration menu

When you select Outcalling Administration from the Voice Administration menu, the Outcalling Administration Menu (Figure 8-35) is displayed.

Figure 8-35xxx
The Outcalling Administration menu

```
                    Outcalling Administration

              1 Outcalling Options

              2 Outcalling Audit Trail Report

 Select an item >

   Exit
```

Procedure 8-37xxx
Using the Outcalling Administration Menu

Starting point : The Voice Administration menu, <4> entered.

The Outcalling Administration menu appears (Figure 8-35).

1 Choose an item by entering its number and pressing <Return>.

The form corresponding to your selection appears. See the following sections for details:

<1> "Outcalling Options"
<2> "Outcalling Audit Trail Report"

2 Select [Exit] to return to the Voice Administration menu.

The Outcalling Options screen

When you select Outcalling Options from the Outcalling Administration menu, the Outcalling Options screen (Figure 8-36) is displayed.

Figure 8-36xxx
The Outcalling Options screen

```
                     Outcalling Administration

  Outcalling Options

  Collect Audit Trail Data:                            No   [Yes]

* Number of days of Audit Data stored:                 7

* Shutdown Audit Trail at Volume Full (Percentage):  85 %

  Maximum Number of Outcalling Channels:               2

  Maximum Number of Remote Notification
  Retry Repeats (before notification to a
  user is disabled by the system):                     5

  Numeric Pager Data Terminator:                       #

**Default Numeric Pager Data:

**Delivery to Non-user on Weekdays from (hh:mm):     00:00 to (hh:mm) 23:59

**Delivery to Non-user on Weekends from (hh:mm):     00:00 to (hh:mm) 23:59

                                                            MORE BELOW
```

```
                     Outcalling Administration          MORE ABOVE
**Delivery to Non-user Retries:
**Busy       Retry limit:  3      Retry Interval (hh:mm):    00:05
**No Answer Retry limit:  10      Retry Interval (hh:mm):    00:15
**Answer     Retry limit:  0      Retry Interval (hh:mm):    00:00

**Delivery to Non-user Addressing Prefixes & Associated dialing Codes
                                  9          9
                                  12

**   Number of times to play a message to a non-user:    2

**   DTMF confirmation overrides user preferences:       No   [Yes]

**Delivery to non-user DTMF Confirmation Required:       [No] Yes

 Select a softkey >
    Save        Cancel
```

 * These fields are displayed only if Collect Audit Trail Data is Yes.
**These fields are displayed only if Voice Messaging (MMUI) is enabled.

The following fields are displayed on the Outcalling Options screen:

- ***Collect Audit Trail Data*** - When this field is set to “Y es” Outcalling Audit Trail Reports are generated by the system. These reports can be used to obtain information about a specific subscriber name, mailbox, or phone number. The reports give you either summary or detailed information about the number of calls, the start time and duration of calls, the numbers called, whether the RN or DNU service was used, and the status of calls. The default is “Yes”. See the section “Outcalling Audit Trail Report” on page 8-132 for more information.
- ***Number of Days of Audit Data Stored*** - This field is displayed if *Collect Audit Trail Data* is set to “Yes”. This field indicates the number of days the audit trail data will be stored on disk before being overwritten. The number of days can range from 1 to 63, with a default of “7”.
- ***Shutdown Audit Trail at Volume Full (Percentage)*** - This field is displayed if *Collect Audit Trail Data* is set to “Yes”. When the volume on which audit trail is stored is almost full, collection of audit trail data is disabled. The value entered here determines the percentage full at which this occurs. (Note that this is a percentage of text space, not voice space.) The default is “85%”.
- ***Maximum Number of Outcalling Channels*** - This field specifies the maximum number of outcalling channels/agents that can be used at any given time by the outcalling service. The default is “2”. This is the recommended maximum for moderate use. For high usage of the outcalling server, the system administrator may have to increase this number. The channels allocated to the OC Service in the Channel Allocation Table (if any), should be entered here.

  Note: Do not allocate all channels to outcalling. For example, if a broadcast message is sent, and many subscribers have RN, all channels could temporarily be taken for outcalling use, leaving no channels available for call answering and message retrieval.

- ***Maximum Number of Remote Notification Retry Repeats*** - This field determines the number of retry cycles or sets allowed before the system disables a subscriber’s Remote Notification feature due to consecutive failures of notification calls. This occurs if the subscriber does not log in and retrieve messages. The default is “5”.

For example, if the system attempts to notify a subscriber of a message, but the notification numbers are not answered, the system will stop notification attempts after the No Answer limit has been exhausted for the subscriber. This is considered one Retry Repeat. If another new message is left for the subscriber, and retry attempts are again exhausted, this would be counted as the second Retry Repeat. This continues until the maximum number of Retry Repeats set in this field is reached, at which time DMS VoiceMail no longer attempts to notify the subscriber of new messages. If a subscriber logs on to the mailbox and retrieves the messages, this parameter is reset to “0”, and outcalling is re-enabled for the subscriber.

- ***Numeric Pager Data Terminator*** - This special digit is required by some general access paging services (such as SkyPager). When DMS VoiceMail calls the paging service and the call is answered, DMS VoiceMail sends the pager identification number (PIN), the terminator digit (using it as a delimiter), then the call-back number, followed by the terminator digit. For example, SkyPager uses “#”. This is also the default. If users subscribe to a general access pager service that does not accept this character, leave this field blank. This will cause DMS VoiceMail to pause several seconds between the PIN and call-back numbers.
- ***Default Numeric Pager Data*** - This field is displayed if: (a) the target DN of any user’s remote notification schedule is that of a general access paging service or if (b) the target DN of any user’s remote notification schedule is that of a numeric pager and the callback number has not been entered in the *Pager Callback Data* field. (See the Outcalling Fields in the Add or Modify Local Voice User screen.)

  The number entered here is displayed on a digital numeric pager and indicates the call-back number to the user. This field works on a system-wide basis so that all users subscribing to a general access paging service will see this number on their pager display. It is also displayed for users with numeric pagers if the callback number has not been defined in the *Pager Callback Data* field in their RN schedules. It is suggested that you enter the external call-back number used to access voice messaging. Up to 8 digits can be displayed.

CAUTION
Geographical restrictions on phone message delivery
In most geographical areas, electronic delivery of phone messages is restricted by law to certain time periods during the day. Administrators are responsible for confirming and implementing the restrictions for their areas. Make sure that you specify the permitted times in the next field.

The following fields are only displayed if Voice Messaging (MMUI) is installed.

- ***Delivery to Non-user Weekdays/Weekends*** - These fields define the time windows during which messages can be delivered to non-users. They are entered in 24-hour clock format and must be chronologically correct. There are two permitted time windows to define: one for Weekdays (Monday to Friday) and one for Weekends. For example, on weekdays you may allow messages to be delivered between 9:00 a.m. and 9:00 p.m. in order not to disturb people in the evening or early morning.

  Note: The default for both weekdays and weekends is “00:00” to “23:59”. Therefore, if you do not modify this field users are allowed to send messages to non-users 24-hours a day, 7 days a week.

  Messages delivered to non-users are subject to *stale dating*. Stale dating is the time period beyond which a delivery attempt will no longer be made. If the stale date is reached and the message is not delivered, a non-delivery notice (NDN) is returned to the originator. The default is 1 day, 12 hours. This parameter is not accessible to you through the DMS VoiceMail administration screens and can only be changed by a representative of your regional support center (RSC).

  When a user sends a DNU message the system checks the current time against the permitted time window to see if it is allowed to send the message. If the current time falls into the restricted time window, the system then checks the stale date parameter to see if the message will have become stale by the time the system is permitted to send messages again.

  The following examples describe possible scenarios to give you an idea of how permitted/restricted time windows interact with stale dating. They all use a permitted time window of 9:00 a.m. to 9:00 p.m. for weekdays.

  Example 1: A user sends a DNU message at 10:00 p.m on a weekday. The stale period is defined as 1 day 12 hours. By 9:00 a.m. the message will only be 11 hours old. The system will send the message at 9:00 a.m. If the call is not answered or busy at that time, the system will use the defined retry limits and intervals (these fields follow this one).

  Example 2: A user sends a DNU message at 10:00 p.m. on a weekday. The stale period has been defined as 10 hours. By 9:00 a.m. the message will be 11 hours old and will have become stale. The system will not be able to send the message in the morning. The system sends a non-delivery notification (NDN) to the user explaining that the message could not be delivered. The NDN will also inform the user of the times during which delivery to non-users is permitted.

Example 3: A user sends a DNU message at 8:30 p.m. The call is not answered. (The *No Answer Retry Limit* is 10 and the *No Answer Retry Interval* is 20 minutes). The system retries the message at 8:50 p.m. (Retry #1). There is still no answer. The system can not retry the message 20 minutes later because this will be within the restricted time period. The stale period is 1 day and 12 hours. The message will be 12.5 hours old by 9:00 am. The system will retry the message at 9:00 a.m. (Retry #2).

- ***DNU Busy Retry Limit*** **-** This field determines the number of times the system attempts to deliver a message to a non-user when the destination number is busy. When this limit is exceeded, the No Answer Retry Limit is used. Therefore, if a number remains busy, the number of call attempts would equal the Busy Retry Limit plus the No Answer Retry Limit. If the No Answer Retry Limit is also exceeded, a non-delivery notification (NDN) is sent to the originator of the message, DNU stops for that message, and the Maximum Number of Retry Repeats is increased by one. You may enter a value from 0 to 10. The default is 3.
- ***DNU Busy Retry Interval*** **-** This field determines the amount of time the system waits before attempting to send the message if the previous attempt was unsuccessful because the destination number was busy. You may enter a value from 00:00 to 23:59. The default is 00:05.
- ***DNU No Answer Retry Limit*** **-** This field determines the number of times the system attempts to deliver a message to a non-user when the destination number is not answered. When the limit is exceeded, a non-delivery notification (NDN) is sent to the originator of the message, DNU stops for this message, and the Maximum Number of Retry Repeats is increased by one. You may enter a value from 0 to 10. The default is 10.
- ***DNU No Answer Retry Interval*** **-** This field determines the amount of time the system waits before attempting to send the message again if the previous attempt was unsuccessful because the destination number was not answered. You may enter a value from 00:00 to 23:59. The default is 00:15.
- ***DNU Answer Retry Limit*** **-** This field determines the number of times the system attempts to deliver a message to a non-user when the destination number is answered but the recipient does not give required DTMF confirmation (by pressing 2 on the telephone keypad). When the limit is exceeded, a non-delivery notification (NDN) is sent to the originator of the message, DNU stops for this message, and the Maximum Number of Retry Repeats is increased by one. You may enter a value from 0 to 10. The default is 0.

Note: If DTMF confirmation is expected, you should not set the Answer Retry Limit higher than one. If the recipient hangs up he probably does not want to hear the message, and DMS VoiceMail should not continue to call him. If the message is delivered to a rotary phone subscriber, the recipient will not be able to press 2, and will become aggravated with repeated attempts to deliver a message.

- ***DNU Answer Retry Interval -*** This field determines the amount of time the system waits before attempting to send the message again if the previous attempt was unsuccessful because the destination number was answered, but the recipient did not provide required DTMF confirmation. You may enter a value from 00:00 to 23:59. The default is 00:00.

  Note: See page 8-131 for an example of a DNU retry scenario.

- ***Delivery to Non-user Addressing Prefixes & Associated Dialing Codes*** When composing a DNU message, the user has to indicate to the system that the address is not that of an internal DMS VoiceMail user or a distribution list number, but that of a non-user outside of the switch. This is done by entering a DNU prefix. When a user enters this number during a compose command the system knows that the number that follows is that of a non-user.

  Note: These prefixes cannot conflict with networking location codes, distribution list numbers, or mailbox numbers. However, conflicts with DNs are allowed.

  You should configure at least two DNU prefixes: one for external numbers and one for internal numbers. The following example illustrates why you might need a DNU prefix for internal numbers. A phone in a meeting room is not likely to be associated with any particular user because it is used as a common phone. However, if a user wants to send a message to this phone, it will have to be sent as a DNU message.

  The DNU prefix for external numbers requires an associated dialing code. When DMS VoiceMail places the call, the prefix is replaced by the associated dialing code which is used to generate the actual phone number that is dialed by the system. It is suggested that the prefixes match the dialing codes whenever possible. For example, a prefix of 9 will be replaced with an actual dialing code of 9 (used by DMS VoiceMail to dial outside of the system). This makes it easier for users as they do not have to remember extra numbers. They simply enter the same number that they dial when calling the person.

The DNU prefix for internal numbers does not require an associated dialing code because DMS VoiceMail does not have to dial out of the system. For example, the extension of the phone in a meeting room is 8001 and the DNU prefix (for internal numbers) is defined as 12. The user enters 128001 and the system dials 8001.

Prefixes can also be used to simplify the dialing process by replacing longer sequences of numbers with a one-digit number or a short number sequence. For example, your users often send messages to numbers in the 513 area code. Enter a prefix, such as 2, and define the dialing code as 91513.

Inform your users of any DNU prefixes that you create.

- ***Number of Times to Play a Message to a Non-user*** - This field determines the number of times a DNU message will be played out to the called party. The default is 2. This is also the maximum value that can be entered.
- ***DTMF Confirmation Overrides User*** - Each subscriber may have chosen to require DTMF confirmation (in User Administration). When this field is set to “Yes” the subscriber’s preference is overrided and the following field, *DNU DTMF Confirmation Required*, takes effect. The default is “Yes”.
- ***DNU DTMF Confirmation Required*** - DTMF confirmation means that a non-subscriber who receives message from a DMS VoiceMail system must press **2** on the telephone keypad to hear messages. When this field is set to “Yes”, all DNU messages (on an system-wide basis) require DTMF confirmation. When set to “No”, all DNU messages are delivered automatically upon voice detection. If you are in an area where rotary phones are widely used, you should leave confirmation disabled. The default is “No”.

DNU Retry Scenario

Busy Retry Limit = 3; Busy Retry Interval = 5 mins
No Answer Retry Limit = 10; No Answer Retry Interval = 15 mins
Answer (2 not pressed) Retry Limit = 0; Answer Interval = 0 mins
DTMF Confirmation is required

Table 8-7xxx
DNU retry scenario

Time of Message	DNU Action	DNU **Result**	Further Action
9:30 a.m. Message 1	DNU message sent	Answered, 2 not pressed	Answer Retry Limit exceeded. There will be no retry attempt.
9:50 a.m. Message 2	DNU message sent	Busy	DNU rescheduled using Busy Retry Limit and Interval
9:52 a.m. Message 3	DNU message sent	No Answer	DNU rescheduled using No Answer Retry Limit and Interval
9:55 a.m.	First Busy retry for message 2	Busy	DNU rescheduled using Busy Retry Limit and Interval
10:00 a.m.	Second Busy retry for message 2	Busy	DNU rescheduled using Busy Retry Limit and Interval
10:05 a.m.	Third Busy retry for message 2	Busy	Busy Retry Limit exhausted; DNU rescheduled using No Answer Retry Limit and Interval
10:07 a.m.	First No Answer retry for message 3	Answered, 2 is pressed	DNU attempts stop for message 3
10:20 a.m.	First No Answer retry for message 2	Answered, 2 is not pressed	Answer Retry Limit exceeded; DNU attempts stop for message 2

Procedure 8-38xxx
Setting Outcalling Options

Starting point : The Outcalling Administration menu, <1> entered.

1 The Outcalling Options form appears (Figure 8-36).

2 Modify the existing information as needed.

3 Select [Save].

The changes are saved and you are returned to the Outcalling Administration menu.

4 Select [Cancel].

You are returned to the Outcalling Administration menu.

Note: *subscriber-specific parameters are defined in the Modify or Add User forms in User Administration*

Outcalling Audit Trail Report

The Outcalling Audit Trail Report form (Figure 8-37) is displayed when you select item <2> in the Outcalling Administration menu. This is a report selection form in which you specify the type of report you want to retrieve (summary or detail). You must specify whether you want to retrieve reports for a particular subscriber, mailbox number, phone number, or all. You can also specify the time period that the report should cover.

Figure 8-37xxx
The Outcalling Audit Trail Report

```
                        Outcalling Administration
Outcalling Audit Trail Report

Report Type:        [Summary]   Detail

Selection Criteria: [All]  Name    Mailbox   Phone_Number

   *  Last Name:            ____________________________________

   *  First Name:           ________________________

   **Mailbox Number:        _________________

   ***Target Phone Number:  ________________________


Report Start(dd/mm/yy hh:mm):____________________ (or blank for oldest)
Report End  (dd/mm/yy hh:mm):____________________ (or blank for newest)
________________________________________________________________________
 Select a softkey >

             Cancel         View         Print
                           Reports      Reports
```

```
* These fields are displayed if Selection Criteria is set to Name.
**This field is displayed if Selection Criteria is set to Mailbox.
***This field is displayed if Selection Criteria is set to Phone Number.
```

The following fields are displayed on the Outcalling Audit Trail Report screen:

- ***Report Type*** - Your options are Summary and Detail. A summary report shows only completed calls. A detail report shows all attempts, both successful and unsuccessful.
- ***Selection Criteria*** - All entries in the database can be viewed or you can view data for a specific subscriber, mailbox number, or phone number.

- ***Last Name*** **-** This field is displayed if *Selection Criteria* is set to Name. If you want to view outcalling data for a particular subscriber, enter that subscriber's full last name (and first name in the next field as there may be more than one subscriber with the same last name). This field accepts all characters except "+", "?", and "_".
- ***First Name*** **-** This field is displayed if *Selection Criteria* is set to Name. If you want to view outcalling data for a particular subscriber, enter that subscriber's full first name (as well as the last name in the previous field). This field accepts all characters except "+", "?" and "_".
- ***Mailbox*** **-** This field is displayed if *Selection Criteria* is set to Mailbox. To view outcalling data for a specific mailbox, enter the full mailbox number. This field accepts all characters, except "+" and "_".
- ***Target Phone Number*** **-** This field is displayed if *Selection Criteria* is set to Phone Number. To view outcalling data for a particular phone number, enter the full number in this field. This field accepts numeric data only.
- ***Report Start/End*** - Enter the start and end date and time to indicate the period of time that should be included in the report. This field accepts numeric data only.

Procedure 8-39xxx
Viewing the Outcalling Audit Trail Report

Starting point : The Outcalling Administration form, <2> entered.

The Outcalling Audit Trail Report appears (Figure 8-37).

1 Change the selection criteria as desired.

2 Use [View Reports].

The audit trail reports are displayed. See the next section, "Outcalling Audit Trail".

3 Use [Cancel].

You are returned to the Outcalling Administration screen.

Procedure 8-40xxx
Printing the Outcalling Audit Trail Report

Starting point : The Outcalling Administration form, <2> entered.

The Outcalling Audit Trail Report appears (Figure 8-37).

1 Change the selection criteria as desired.

2 Ensure that the printer is on-line and has paper.

3 Use [Print Reports]. (Ensure that the printer is on-line.)

A new set of softkeys are displayed: [Cancel Printing] and [Continue Printing].

4 Use [Continue Printing] to print the report or [Cancel] if you do not want to print the report.

If you selected [Continue Printing], a [Cancel] softkey is displayed which can be used to cancel printing once printing has started.

You are returned to the Outcalling Administration screen.

Outcalling Audit Trail

The Outcalling Audit Trail Report is displayed when you use the [View Reports] softkey on the Outcalling Audit Trail form. The format of the report is either in summary or detail mode, depending on the selection made on the Outcalling Audit Trail form. Figure 8-38 shows a summary report. Figure 8-39 shows a detail report.

Figure 8-38xxx
The Summary Outcalling Audit Trail Report

```
                       Outcalling Administration

Outcalling Audit Trail from 01/10/90 to end of data.

Date (dd/mm/yy)
Name           Mailbox Number
Start        Duration  Phone Number           Type    Call Status
(hh:mm)      (mmm:ss)

10/01/90
Smith, J       7550
12:40        1:10      98292962               DNU     Answered
12:45        0:05      98292962               DNU     No DTMF Conf.
13:45        0:18      8051-345643            RN      Answered

10/02/90
Jones, D       7091
8:52         0:02      8052                   RN      Answered
8:57         0:06      8052                   RN      Disabled

Select an item >
   Exit                                  Next Page
```

The summary report displays the following information:

- ***Date*** **-** The date on which the call was made.
- ***Name*** **-** The name of the DMS V oiceMail subscriber who initiated the call.
- ***Mailbox Number*** **-** The mailbox that originated the call.
- ***Start Time*** - The time at which the call was answered.
- ***Duration*** - The length of the call in minutes and seconds.

- ***Phone Number*** - The number called. A maximum of 30 digits can be displayed in this field. For calls placed to paging services (such as SkyPager), the PIN number is also displayed (e.g., in 8051-345643, the last 6 digits are the PIN number). If the full number is longer than 30 digits, the first few digits in the paging service phone number will be truncated.
- ***Type*** - This field displays the outcalling service that was used: either Remote Notification or Delivery to Non-user.
- ***Status*** - This field displays the result of the call.
  - ***Answered*** - indicates that the destination number was answered and the message was heard by the called party.
  - ***RN Disabled*** - indicates that the called party answered and pressed 3 to disable RN.
  - ***No DTMF Confirmation*** - indicates that the called party did not press 2 to hear a DNU message (not relevant if DTMF confirmation is not required).

Figure 8-39xxx
The Detail Outcalling Audit Trail Report

```
                       Outcalling Administration

Outcalling Audit Trail from 10/10/90 to end of data.

Date (dd/mm/yy)
Name      Mailbox Number
Transaction    Start  Duration  Device/Target Phone Number Channel  Re-
(hh:mm)       (hh:mm) (mmm:ss)                                 DN    try
Request #    Outcall Process     Call Status    Outcall Action

10/02/90
Howe G.   3000
15:10          15:10
#1137        RN Submission                      Continue             0
15:10          15:10
#1138        RN Validation                      Continue             0
15:10          15:10  0:15      PaSrv/8051-345643            2004    0
#1139        RN Call Results     Answered       Remove,user logged in 0

Select a softkey>
  Exit                                          Next
                                                Page
```

The additional fields on the detailed Audit Trail Report are:

- ***Transaction Time*** **-** This field indicates the time at which the delivery should have taken place.
- ***Start Time*** - The time at which the current outcall process started.
- ***Duration Time*** - The length of the call.
- ***Device/Target Phone Number*** - The type of device called followed by the phone/pager number. The device will be one of the following:
  - Phone
  - ToneP (tone pager)
  - V oice (voice pager)
  - NumPa (numeric pager)
  - PaSrv (pager service)

  If the device is a phone, the device type is followed by the number that was called. If the device is a pager service, the device type is followed by the pager service number and the pager identification number (PIN). The maximum length for this field is 30 digits. If this limit is exceeded, the first few digits of the paging service phone number will be truncated.
- ***Channel DN*** - The DN associated with the voice channel used.
- ***Retry*** **-** Combined retry count at the time of the attempt.
- ***Transaction Request Number*** - A unique number identifying the (RN or DNU) request.
- ***Outcall Process*** **-** The type of audit trail entry. This could be:
  - ***Submission*** - indicating that a request has been made for an outcalling service.

  Instead of “Submission” you may also see one of the following: “Recovery” or “Logout/Admin”. They are considered special cases.

  - ***Recovery*** - indicates that messages for outcalling have been detected and submitted after a system reboot.
  - ***Logout/Admin*** - indicates that one of two conditions has occurred. The first possibility is that a user has logged out with unannounced messages left in their mailbox. Normally, if a user is listening to a message when a new message comes in the new message is announced after the user has finished listening to the other message. However if the user hangs up before the message has finished playing, the new message will not be announced. (In this situation the user will continue to be notified of messages.). The second possibility is that the administrator has modified a user’s account while there are unread messages in the user’s mailbox.
  - ***V alidation*** - indicates a checking process just before a call was/is made;

- *Call Results* - indicates information regarding the Call Status and Outcall Action in the adjacent fields.

- ***Call Status*** - This is a general statement of the results of a call. The possibilities are: “Busy”, “Answered”, “No Answer”, “No DTMF Conf”, “Reorder”, “Resource Delay”. Resource Delay is entered when a channel attempting an outgoing call is interrupted by an incoming call, which is given priority. The outgoing call is retried on a different channel.
- ***Outcall Action*** - This field indicates the action performed on the request. The possibilities are:
  - ***Continue*** - The validation has been passed and a call attempt is to be made.
  - ***Remove*** (retry limit reached) - After the call, the retry was not rescheduled because the retry limit had been reached.
  - ***Remove*** (another RN exists) - The validation step determined that the user has logged on since the last RN attempt and the retry was cancelled.
  - ***Defer*** - Another call attempt has been scheduled.

    RN calls to pagers are always rescheduled (providing that the retry limit has not been reached) because a call to a pager is never a “success”. The user has to log on to his or her mailbox when the notification is received (i.e., on the same call) for the call to be considered a success. Note that if the user logs on before the next retry, the retry will be cancelled.

Procedure 8-41xxx
Viewing Outcalling Audit Trail Reports

Starting point : The Outcalling Audit Trail Report, [View Reports] entered.

1 The Outcalling Audit Trail forms appears.

2 Use [Next Page]. (When the reports are finished, a prompt appears indicating it is the end of the report.)

The pages of the reports are displayed.

3 Use [Exit].

You are returned to the Outcalling Audit Trail Reports screen.

Voice Form Definitions

Note 1: This feature is optional and may not be installed on your system.

Note 2: Voice Forms can not be enabled for systems that have the Call Answering (VMUIF) feature installed.

Overview

Meridian Voice Forms allow you to create and maintain customized information gathering applications. The applications you create can be viewed as the electronic equivalent of the traditional printed form or questionnaire. Voice forms offer your organization a convenient way of collecting information. They also enhance your ability to reach customers and potential clients by making it convenient for them to reach your organization, twenty-four hours a day, from any place. Furthermore, clients are no longer required to perform the often tedious task of filling out a traditional paper questionnaire or form.

The Meridian Voice Forms feature is documented in two other documents:

- the *Meridian Voice Forms Implementation Guide*
- the *Meridian Voice Forms Transcriber User Guide*

Make sure you have all of these guides on hand before you begin planning and configuring your voice form applications.

Begin by reading the Meridian Voice Forms Implementation Guide. This guide will help you plan and design your voice forms. It includes worksheets which you can fill out while designing the voice form on paper. Once you are ready to configure the voice form in DMS VoiceMail, simply copy the information into the various administration screens that are used to configure voice forms.

When you are ready to configure your voice form, return to this section which provides detailed information about configuring voice forms in DMS VoiceMail.

Then read the Meridian Voice Forms Transcriber User Guide. As the administrator, you will have to be familiar with this process so that you can test the transcriber interface for each voice form and so that you can train your transcribers.

Use Table 8-8 to locate the appropriate document for a particular step in the process of creating a voice form application. It is assumed that voice forms have been installed on the system.

Table 8-8xxx
Creating a voice form application

Step	Description	Document/Section
1	Familiarize yourself with the feature.	*Implementation Guide*: “Introduction”, “Overview of Meridian Voice Forms”, “Scenarios”
2	Plan and design your voice form application.	*Implementation Guide*: “Creating Voice Form Applications”: “Recognizing the need for a voice form application”, “Designing the application”
3	Sequence and/or flowchart the application.	*Implementation Guide*: “Sequence and/or flowchart the application:
4	Identify the general characteristics of the form.	Read the following section in the *Administration Guide*: “Defining form characteristics” for an explanation of the form characteristics that you will be defining. Then fill out the Voice Form Definition Worksheet in the *Implementation Guide*. (One for each voice form.)
5	Identify the fields within the form.	Read the following section in the *Administration Guide*: “The Insert New Field screen”. Then fill out a New Field Worksheet for each field in the form. Note that there are two different worksheets: one for No Answer fields, and one for Voice Answer fields. These are both in the *Implementation Guide*. Assemble the worksheets so that they represent the order in which the fields will be presented in the form.
6	Configure the voice form application in DMS VoiceMail.	*Administration Guide*: From “Viewing a list of existing voice forms: the Voice Forms screen” to “Saving the voice form definition”
7	Script and record the prompts.	*Implementation Guide: “Scripting the prompts” and “Recording the prompts”* *Administration Guide*: “Recording field prompts and field names”
8	Build the service through which the voice form will be accessed.	*Administration Guide*: “Making voice forms accessible”
9	Define the transcription service DN.	*Administration Guide*: “Defining a VSDN for the transcription service”
10	Test the voice form by calling it and recording a response.	Implementation Guide: “Testing voice forms”
11	Test the transcription service by transcribing your test response.	Meridian Voice Forms Transcriber User Guide.
12	Modify the voice form if necessary.	*Administration Guide*: “Modifying or viewing a voice form”
12	Train your transcribers	*Implementation Guide:* “Provide training for transcribers”
13	Maintain and manage the voice form.	Implementation Guide: *“Maintaining voice forms” and “Managing voice forms”*

Viewing a list of existing voice forms: the Voice Forms screen

When you select Voice Form Definitions from the Voice Administration menu, the first screen displayed is the Voice Forms screen. If you have just installed voice forms and have not yet created any voice form applications, this screen will appear as depicted in Figure 8-40. Once you have created at least one voice form, additional softkeys appear at the bottom of the screen, as shown in Figure 8-41. This screen presents a complete list of the voice form definitions currently residing on the system.

Figure 8-40xxx
The Voice Forms screen (before any voice forms are created)

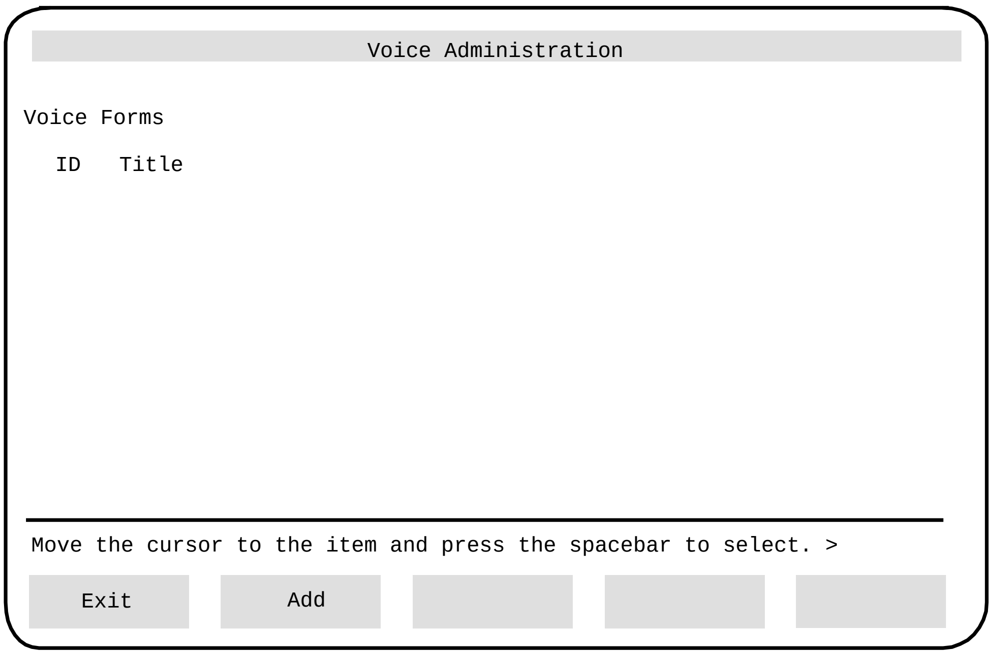


Figure 8-41xxx
The Voice Forms screen (voice forms have been created)

```
                         Voice Administration

Voice Forms

  ID   Title
  01   Automatic Deposit Authorization
  02   Credit Card Application
  03   Consumer Loans
  04   Small Business Loans
  05   Car Loans
  06   Consumer Overdraft Protection Plan
  07   Pre-Authorized 1st Mortgage
  08   Application for 2nd Mortgage
  09   Application for Convenience Card
  10   Pre-Authorized Payment Plans
  11   Address Change
  12   Personal Cheque Orders              * Incomplete Recordings

Move the cursor to the item and press the spacebar to select. >

   Exit        Add/Copy      View/Modify     Delete
```

The following information is displayed for each existing voice form definition:

- The form ID number. This number is defined when you add a voice form definition.
- The title of the voice form application.
- A status flag, if necessary, to indicate incomplete recordings. This flag is displayed if a voice form definition has been saved with one or more of its required field prompts or field names left unrecorded or if the form does not contain at least one field.

  Note: A voice form application cannot be used if there are incomplete recordings.

You may have to check the Voice Forms list from time to time to keep yourself updated on the current application load, as well as to verify which forms are incomplete, and therefore non-functional.

Procedure 8-42xxx
Accessing the Voice Forms screen

Starting Point The Main Menu.

1 Select Voice Administration <Return>.

2 Select Voice Form Definitions <Return>

The Voice Forms screen is displayed.

From this screen it is possible to add a new voice form definition, copy from an existing definition, view an existing definition, modify an existing definition, delete an existing definition, or return to the Voice Administration menu. These actions are described in the following pages.

Adding a new voice form definition

If you are creating your first voice form, or if the required application is significantly different from other existing voice form applications, you will need to add a new voice form definition. This creates a file which contains all of the information that is relevant to the particular voice form application.

The Add a Voice Form Definition screen

New voice form definitions are added to the system using the Add a Voice Form Definition screen (Figure 8-42). The fields in this screen define the general structure of the voice form application.

The procedure for adding a voice form definition is different if you are adding the very first voice form to the system. Follow Procedure 8-43 to add the first voice form to the system. For all subsequent voice forms, follow Procedure 8-44.

Procedure 8-43xxx
Adding the first voice form definition

Starting point The Voice Forms screen.

1 Press the [Add] softkey.

The Add a Voice Form Definition screen is displayed (see Figure 8-42).

Procedure 8-44xxx
Adding subsequent voice form definitions

Starting point The Voice Forms screen.

1 Press the [Add/Copy] softkey.

2 A new set of softkeys is displayed as shown below.

```
Select a softkey >
            Cancel      Add New     Copy Form
                        Form
```

3 Select [Add New Form].

The Add a Voice Form Definition screen is displayed (see Figure 8-42).

Defining form characteristics

The Add a Voice Form Definition screen is the starting point for voice form definition. It is where you define the general, caller, and transcriber characteristics that will affect how the form will function.

Figure 8-42xxx
The Add a Voice Form Definition screen

```
                          Voice Administration
Add a Voice Form Definition

Voice Form ID:                        02
Title:                                Credit Card Application
Form Name Recorded (Voice):           No
Transcription Password:               6060
Maximum Untranscribed Responses:      1000
Overflow Handling DN:                 8050
New Responses Notification DN:        8051
New Responses SMDI Link Name DN:      _____________
Special Responses Notification DN:    8052
Special Responses SMDI Link Name:     ____________
                                                     MORE BELOW
select a softkey>
   Save        Cancel     Open/Modify     Voice
                          Fields
```

```
                          Voice Administration           MORE ABOVE
Add a Voice Form Definition

Transcription Field Separator:        [Field Name]  Tone  Silence
Default Field Separator Delay:        Stop  [Delay]  deciseconds: 0
Play Envelope for Header:             [No]  Yes
Delay After Header (deciseconds):     30
Caller Confirmation Mode:             [None]  At_Each_Field  Whole_Form
Default Revert DN:                    0
Caller '0' Allowed                    [No]  Yes
*System Messages File:                [English]
                                      French

select a softkey>
   Save        Cancel     Open/Modify     Voice
                          Fields
```

*This field is displayed only if more than one language is installed.

Fill in the following fields as required. Get the Voice Form Definition Worksheet from the *Meridian Voice Forms Implementation Guide* and copy the information from the worksheet into the Add a Voice Form Definition screen.

- ***Voice Form ID -*** This is the identification number that uniquely identifies this voice form definition. The ID number assigned must be unique across all voice service applications that currently reside in the system (voice menus, applications, thru-dialers and time-of-day controllers). If you assign a duplicate ID, an error message will be displayed on the command line of the administration terminal. The voice form definition cannot be saved until you assign a unique ID number to it.

  Come up with some sort of numbering scheme so that when you look at an ID you will know that it belongs to a voice form. For example, if all of your voice menu applications are numbered 3333xxxx, you may want to number your voice forms 4444xxxx.

- ***Title -*** The name assigned to the voice form. Ensure that it is descriptive enough of the form's purpose to allow you to easily recognize which voice form application you are dealing with.

  Note: If you have created this voice form definition by copying from an existing definition, it is not necessary to provide a new title in order to save the definition. However, it is recommended that you do so. This will avoid any possible confusion caused by having two voice form definitions with the same title.

- ***Form Name Recorded (Voice) -*** This field indicates whether or not a voice recording of the form's name has been created. This name is played to transcribers only and helps them to identify the form. To review the current name (if there is one) or record a name press the [Voice] softkey while the cursor is in this field. See the chapter "Making recordings" if you do not know how to create a recording using the [Voice] softkey.

  Modifying an existing form name does not affect the availability of existing caller responses for transcription. However, you should notify transcribers about your intention prior to changing the name. This will avoid any confusion that could occur when the new form name is announced to transcribers expecting to hear the old form name.

  If you do not record a voice form name, transcribers will hear the voice form ID instead.

- ***Transcription Password -*** This password is optional. If you enter a password in this field, transcribers trying to access the voice form will have to enter this password before being allowed to listen to caller responses. You may enter a password of 1 to 16 numeric characters in length (no spaces allowed). If you do not want transcribers to have to enter a password, leave this field blank.
- ***Maximum Untranscribed Responses -*** This field indicates the maximum number of caller responses that are allowed to exist at any one time for this voice form application. It is not recommended that you enter a value greater than 1000 (although this field will accept higher values).

  If you will be creating a lot of voice form applications, be careful of how high you set this value. As the number of untranscribed responses is allowed to accumulate, the amount of free disk space will decrease significantly and you may eventually reach full capacity. Prioritize your voice forms in terms of the importance of the responses you are expecting. Assign higher values to your more important applications to ensure that no responses are lost. For example, forms that are designed to take orders or record customer complaints and/or problems may be considered more important than customer surveys and probably warrant a higher value. In general, you should advise your transcribers to retrieve new responses as soon as possible.

  You can also use this field to limit caller responses to a specific number. A good example is a phone-in giveaway in which the first 100 callers receive some sort of complementary gift. In this situation you could enter 100 as the maximum value so that only the first 100 caller responses are recorded.
- ***Overflow Handling DN -*** This field is used when the maximum number of untranscribed responses is reached. If the limit has been reached or surpassed, callers who try to access the voice form are transferred to the DN specified in this field. The DN may be from 1 to 30 characters in length using the following characters: 0-9, #, *, (,), -, _, $ (however, the DN cannot start with '$').

  If this field is left blank, the following prompt is played to callers who reach the voice form when the maximum number of untranscribed responses has been recorded: *"No more responses can be recorded. Your call cannot be completed at this time. Please try again later. Goodbye."*

- ***New Responses Notification DN -*** This field specifies the DN to which notification is sent when new caller responses have been received and are awaiting transcription. See the section “Notification DNs” in the *Meridian Voice Forms Implementation Guide*. This section describes four different methods for notifying transcribers. If you use the fourth method, which does not use the notification feature, leave this field blank. If you do enter a DN in this field, make sure that it is unique across all notification DNs in the system. The DN may be from 1 to 30 characters in length using the following characters: 0-9, #, *, (,), -, _, $ (however, the DN cannot start with ‘$’).
- ***New Responses SMDI Link Name -*** This field indicates the name of the Simplified Message Desk Interface link in the hardware database which should be used for setting the MWI for new responses. Check the SMDI data port screen using Hardware Administration to determine the setting.
- ***Special Responses Notification DN -*** This field specifies the DN to which notification is sent to indicate that caller responses, marked as “Special” by transcribers, exist. Responses may be marked special if they could not be transcribed for some reason yet need to be taken out of the list of new responses. For example, the response may have been given in a language that the transcriber could not translate.

  See the section “Notification DNs” in the *Meridian Voice Forms Implementation Guide*. If you enter a DN in this field, make sure that it is unique across all notification DNs in the system. If transcribers will be notifying the supervisor of special responses verbally, leave this field blank. The DN may be from 1 to 30 characters in length using the following characters: 0-9, #, *, (,), -, _, $ (however, the DN cannot start with ‘$’).
- ***Special Responses SMDI Link Name -*** This field indicates the name of the Simplified Message Desk Interface link in the hardware database which should be used for setting the MWI for special responses. Check the SMDI data port screen using Hardware Administration to determine the setting.
- ***Transcription Field Separator -*** This field defines what a transcriber hears immediately before an answer is played back. Your choices are:
  - ***Field Name -*** A recording of the field name precedes the answer. The actual name is recorded in the Insert New Field or View/Modify Field screen using the [Voice] softkey.
  - ***Tone -*** The transcriber hears a short tone before hearing the answer.
  - ***Silence -*** Each answer is preceded by a brief period of silence.
- ***Default Field Separator Delay -*** This field determines whether playback stops after each answer or if there is a delay of a prescribed amount of time. The purpose of stopping or delaying after each answer is to give the transcriber enough time to transcribe answers.

- ***Stop -*** The voice form stops playback after each answer is played. The transcriber must use the Play or Skip Forward command to go to the next answer.
- ***Delay (deciseconds) -*** A delay (of the length specified in this field) follows each answer. Specify the length in deciseconds. This delay can range from 0 to 32767 deciseconds.

  A decisecond equals one tenth (1/10th) of a second. Therefore, to determine the number of deciseconds required, decide on the required duration of the delay in seconds and then multiply this value by 10. For example, to configure a delay of 15 seconds, enter 150 in this field.

Note: This parameter is also configurable for each field in the voice form. The value entered here is used as a default when new voice form fields are created.

- ***Play Envelope for Header -*** This field determines whether transcribers hear a full response envelope (“Yes”) or a standard response header (“No”) each time they move to a different caller response. The standard header contains the status (New, Special, or Deleted) and the response number. The full header envelope contains the status, response number, form ID or form name and the date and time the response was recorded.
- ***Delay After Header (deciseconds) -*** This field specifies the amount of delay presented to a transcriber after a response header is played. This delay is useful if part of the header needs to be transcribed. The delay can range from 0 to 32767 deciseconds. (1 decisecond = 1/10th of a second.)
- ***Caller Confirmation Mode -*** This field specifies when callers are asked to confirm their recorded answers (if at all). Confirmation allows callers to hear their answers played back. After an answer is played back, the caller is prompted to accept the answer or to change the answer.

  This field interacts with another field, *Field to be confirmed*, in the Insert New Field screen (Figure 8-45). The manner of interaction is described for each option below:

  - ***None -*** Callers are not asked to confirm their answers. The *Field to be confirmed* field is not displayed in the Insert New Field screen.
  - ***At Each Field -*** For every voice form field that you create, you will specify if a confirmation is required by setting the *Field to be confirmed* field to “Yes” or “No”. This means that you can selectively choose which fields require confirmation. When a field requires confirmation and the confirmation mode is “At Each Field”, the caller is asked to confirm their response immediately after recording an answer for that field.

- ***Whole Form -*** The caller is asked to confirm his or her answers upon completion of the form. A list of answers to those fields requiring confirmation (i.e., those fields for which *Field to be confirmed* is set to “Yes”) is played back for sequential confirmation.

If you select “Whole Form” or “At Each Field” you may want to tell callers something about answer confirmation in an instructional prompt. You should inform when they will be given the chance to confirm their answers and that they can re-record an answer up to three times. You should also mention that only callers with touch-tone phones will have this capability.

- ***Default Revert DN -*** This field defines the DN to which callers are transferred if there is a problem accessing the voice form or if they explicitly ask to be transferred to an attendant by pressing “0”. The DN may be from 1 to 30 characters in length using the following characters: 0-9, #, *, (,), -, _, $ (however, the DN cannot start with ‘$’).

  A revert DN can also be configured for each field in the voice form. The DN entered here is used as a default for each new voice form field that is created.

- ***Caller ‘0’ Allowed -*** When this field is set to “Y es” callers are permitted to press “0” to transfer to an attendant. (Callers are transferred to the DN specified in the *Revert DN* field.) When set to “No” callers are not permitted to do so. (If this field is set to “No” and a caller does press “0”, a message is played indicating that no operator is available. They have the option of continuing with the voice form or cancelling their call.)

  Note: If you allow callers to use “0”, you will have to indicate this in one of the form’s prompts, otherwise callers will not be aware of this capability. You may want to include this in the introductory greeting. You should also indicate that this capability is only available to touch-tone phone users and not to rotary phone users.

- ***System Messages File -*** This field is only displayed if more than one language is installed. The selection made here determines the language in which system prompts are played to callers and transcribers using this voice form. This field does not affect your custom recordings.

Defining voice form fields

Once the general characteristics of the voice form application have been defined, you are ready to add fields. Each greeting, instruction, prompt and question that is to be included in the voice form application requires a field. The field definition that you create contains the actual recorded prompt, the recorded field name which is played back to transcribers and to callers during confirmation and characteristics that are specific to each field.

Opening and modifying fields

When you press the [Open/Modify Fields] softkey, the List of Fields screen for that voice form definition is displayed. If you have not defined any fields, the screen will appear as shown in Figure 8-43. The only entry in this screen is the <END-OF-FORM> marker in the Type column. If you have selected “whole-form” as the confirmation mode, the <CONFIRMATION> marker will precede the <END-OF-FORM> marker. If at least one field has been defined, the screen will appear as shown in Figure 8-44. This screen displays all of the fields that are currently defined for this voice form definition.

Figure 8-43xxx
The List of Fields screen (no voice forms defined)

```
 Credit Union             Voice Administration


List of Fields   Form ID: 02    Title: Credit Card Application

  Field Name        Type           Prompt Rec. Name Rec. Modified
                    <CONFIRMATION>*
                    <END-OF-FORM>



Move selection bar to a field, then select a softkey>

  Close        Insert New
  Fields       Field
```

```
* The CONFIRMATION marker is displayed only if Confirmation Mode is
  set to Whole Form.
```

Figure 8-44xxx
The List of Fields screen (existing voice forms)

```
 Credit Union                Voice Administration


List of Fields   Form ID: 02    Title: Credit Card Application

  Field Name        Type           Prompt Rec. Name Rec. Modified
  Welcome           No Answer      Yes
  Name              Voice Answer   Yes          Yes
  S.I.N.            Voice Answer   Yes          Yes
  Street Address    Voice Answer   Yes          Yes
  Apt. Number       Voice Answer   Yes          Yes
  City/Province     Voice Answer   Yes          Yes
  Postal Code       Voice Answer   Yes          Yes
  Phone number      Voice Answer   Yes          Yes
  Years at address Voice Answer   Yes          Yes
  Employer          Voice Answer   Yes          yes
  Street Address    Voice Answer   Yes          Yes
  City/Province     Voice Answer   Yes          Yes
  Postal Code       Voice Answer   Yes          Yes
  Phone Number      Voice Answer   Yes          Yes
  Years Employed    Voice Answer   Yes          Yes
                    <CONFIRMATION>*
                    <END-OF-FORM>
________________________________________________________________________
Move selection bar to a field, then select a softkey>

  Close        Insert New    View/Modify    Delete       Move
  Fields       Field         Field          Field        Field
```

* The CONFIRMATION marker is displayed only if Confirmation Mode is Whole Form.

The following read-only fields are displayed:

- ***Form ID -*** serves to remind you of the voice form definition you have selected to modify.
- ***Title -*** serves to remind you of the voice form definition you have selected to modify.
- ***Field Name -*** The name of the existing field.
- ***Type -*** This field indicates whether or not an answer is expected for the field. There are, therefore, two types of fields:
  - ***No Answer*** fields only play a prompt. No answer is expected from the caller.
  - ***Voice Answer*** fields play a prompt and record an answer.
- ***Prompt Rec -*** This field indicates whether or not a voice prompt has been recorded for the associated field. (A voice prompt can be a welcome or farewell greeting, a question or an instruction.) If there are incomplete recordings, i.e., fields for which there is no prompt recording, callers will not be able to use the voice form.

- ***Name Rec* -** This field indicates whether or not a recording of the field name has been made. Only “Voice Answer” fields can have field names. This name is announced to callers during confirmation and to transcribers during transcription.
- ***Modified* -** An asterisk (*) in this field indicates that the field has been modified at any time since this voice form has been opened for editing.

Markers

There are two markers that can appear in the *Type* field.

<END OF FORM>

This marker is always present. It is the final entry in the *Type* field and indicates the end of the list of fields for this form. All of the fields that you create will appear above this marker.

<CONFIRMATION>

This marker only appears if you set the Caller Confirmation mode to “Whole Form” in the Add a Voice Form Definition screen.

It indicates the place at which the caller will be asked to confirm his or her answers. By default, this marker appears just before the <END-OF-FORM> marker, but you can move it using the [Move Field] softkey. This may be necessary if you want to include a thank you or farewell prompt just before the caller is disconnected from the form (and therefore, after confirmation). Or, after the caller confirms his or her answers, you may want to include a prompt that asks callers to leave comments of a more general nature. See the section “Moving a field” for more information.

If “At Each Field” was selected as the confirmation mode, callers are asked to confirm an answer immediately after it is recorded. You will not, however, see the <CONFIRMATION> marker in the List of Fields screen.

Defining a new field

If you are creating a voice form from scratch, you will have to define all of the fields that are to make up the application. New fields are added to a voice form definition using the [Insert New Field] softkey in the List of Fields screen. Once you have created a voice form, you can always return to it and insert more fields if required. (See “Modifying a voice form definition” for details.)

Before pressing the [Insert New Field] softkey, make sure the cursor is in the correct location. When you insert a new field, it is placed in the row above the cursor.

Note 1: Get the following worksheets: the New Field Worksheets for “No Answer” Fields and the New Field Worksheets for Voice Answer Fields that you filled out while planning your voice form using the *Meridian Voice Forms Implementation Guide*. You will have one worksheet per field. Simply enter the values from these worksheets into the Insert a New Field screen.

Note 2: You can define up to 150 fields for a single voice form application.

Procedure 8-45xxx
Inserting a new field

Starting point The List of Fields screen.

1 If this is the first field to be added to the voice form. go to step 1a. For subsequent fields, go to step 1b.

a. Position the cursor on the <CONFIRMATION> marker (if it is present). If there is no <CONFIRMATION> marker, position the cursor on the <END-OF-FORM> marker.

b. Position the cursor to indicate where the new row should be inserted. The new row will be inserted above the row in which the cursor is placed. When you are ready to insert the last field in the form, position the cursor on the <END-OF-FORM> marker.

2 Press the <space bar> to select the row.

3 Press [Insert New Field].

The Insert New Field screen is displayed (see Figure 8-45).

4 Select the Field Type.

Your selection will affect the screen display. See Figure 8-45 for “Voice Answer” fields. See Figure 8-46 for “No Answer” fields.

5 Fill in the required information (according to your New Field worksheets). The fields in this screen are described on the following pages.

Record the field name. You can either record the field prompt now (for shorter applications) or during a separate recording session (for longer applications). Follow Procedure 8-46 on page 8-162 to record a prompt.

6 To save the field definition, go to step 6a. If you do not want to save the field definition, go to step 6b.

a. Press [Save].

If all of the mandatory fields have been filled in, the field definition is saved. If something is missing from your definition, you will be informed and the field definition will not be saved until all of the required information has been entered.

Note: *When you return to the Add a Voice Form Definition, be sure to use the [Save] softkey that is displayed. If you do not save the voice form, the fields you have just created will not be saved.*

b. Press [Cancel].

Any changes that have been made are discarded. The List of Fields screen is displayed.

Note: When you save a voice form field, you are returned to the List of Fields screen and the cursor is automatically positioned on the first row. This means that you will have to cursor down to the field you just finished defining before you insert the next field.

The Insert New Field screen

Each field in the voice form is configured at the Insert New Field screen. If the field type is “Voice Answer”, the screen will appear as shown in Figure 8-45. If the field is “No Answer”, the screen will appear as shown in Figure 8-46.

Figure 8-45xxx
The Insert New Field screen for a “Voice Answer” type field

```
                              Voice Administration
Insert New Field    Form ID: 02      Title: Credit Card Application

Field Type:                        No-Answer  [Voice-Answer]

Field Name:                        Welcome_______________

Field Prompt Recorded (Voice):     No

Field Name Recorded (Voice):       No

Answer Length Limit (seconds):     46_

No Answer Handling:                Next Field  Revert Disconnect [Repeat]

*Revert DN:                        0___

**Repeats Before Disconnect:       1

Stop Recording on Silence:         No   [Yes]

***Field to be Confirmed:          No   [Yes]

Save Response if Hangup:           No   [Yes]

Field Separator Delay:             Stop  [Delay]  deciseconds:____
__________________________________________________________________________
Move selection bar to a field, then select a softkey>

  Save         Cancel                     Voice
```

```
*  This field is displayed only if No Answer Handling is Revert.
** This field is displayed only if No Answer Handling is Repeat
***This field is displayed only if Confirmation Mode is Whole-Form or
   At-Each-Field.
```

The following fields are read-only:

- ***Form ID -*** is a read-only field. It serves to remind you of the voice form application you are currently working on.
- ***Title -*** is a read-only field. It serves to remind you of the voice form application you are currently working on.

Fill in the following fields for a “Voice Answer” field.

- ***Field Type -*** specifies the type of voice form field. The selection you make here will affect which fields are displayed on this screen. Your choices are:
  - ***No Answer -*** This type of field plays a prompt (a greeting or instruction) to the caller. The caller simply listens to the prompt without responding.
  - ***V oice Answer -*** This type of field plays a prompt (a question) to the caller. The caller is typically expected to respond. (However, not all questions necessarily require a response. This will depend on how the *No Answer Handling* field is configured.)
- ***Field Name -*** specifies the name of the field. This name should be unique and descriptive enough to identify the function of the field. You may enter a name that is between 1 and 29 characters in length. This is a mandatory field.
- ***Field Prompt Recorded (Voice) -*** indicates whether or not a prompt has been recorded for this field. This prompt is heard by the caller as soon as they reach this field in the voice form. The prompt usually contains information (for No-Answer fields) or a question (for Voice-Answer fields). To record a voice prompt, press the [Voice] softkey while the cursor is positioned on this field. See the section "Recording field prompts and field names" on page 8-162 for more information.

  Note: All field prompts must be recorded before the voice form can be used.

- ***Field Name Recorded (Voice) -*** This field indicates whether or not a recording of the field's name has been recorded. This is a short spoken name, identifying the field (such as "Name", or "Home Phone Number") and is heard by callers during whole-form confirmation and by transcribers if "Field Name" is specified as the Transcription Field Separator. To record a field name, press the [Voice] softkey while the cursor is positioned on this field. See the section "Recording field prompts and field names" on page 8-162 for more information.

  Note: All Field Names must be recorded before the voice form can be used.

- ***Answer Length Limit*** **-** This field specifies the maximum recording length (in seconds) allowed for a caller's answer to the prompt recorded for this field. The minimum value is 1 second and the default maximum value is 30 seconds. This upper limit can be changed by a representative of your Northern Telecom support organization (and may have already been changed during installation). Use shorter limits for yes/no answers and longer limits for answers that require more detailed information. If you feel that the question can not be answered in the maximum time allowed, reword it so that it asks for more specific information or break it into several questions if possible. If you find that this is the case for many of the questions you are including in voice forms, call your Northern Telecom support organization and ask to have this upper limit changed.
- ***No Answer Handling*** **-** This field determines what happens if the caller does not record an answer for this field. Your options are:
  - ***NextField*** **-** The recorded prompt in the next field is played to the caller. Transcribers will hear the message "No answer was recorded" when this field is reached during transcription.
  - ***Revert*** **-** The caller is transferred to the number specified in the *Revert DN* field. This could be the DN of an operator designated to handle calls from people having trouble filling in forms or a transcriber if he or she is usually situated at his or her telephone set.
  - ***Disconnect*** **-** The call is disconnected. The system plays *"Goodbye"* before doing so. (Don't mistake this with the Action After Field disconnect.) This action may be selected for high-volume voice forms where it is important that callers do not tie up lines or if all questions must be answered for the form to be processed. Otherwise, this option is not recommended.

    If during testing, or while the form is in use, it is noticed that callers are being disconnected unexpectedly, it may be because "Disconnect" was unintentionally selected for a field. This is the first thing to check should this occur.
  - ***Repeat*** **-** The field prompt is repeated. If the caller still fails to record an answer, the field prompt will be played again. This continues until the number of repeats specified in the *Repeats Before Disconnect* field has been reached. Once this limit is reached, the caller is disconnected from the voice form. The next field is **not** played.
- ***Revert DN*** **-** This field is only displayed if the *No Answer Handling* field is set to "Revert". This is the DN to which a caller is reverted if he or she does not record an answer.

- ***Repeats Before Disconnect -*** This field is only displayed if the *No Answer Handling* field is set to “Repeat”. The field defines the number of times the field prompt is repeated when a caller does not record an answer to the prompt. After the specified number of repeats is reached, a system ‘goodbye’ message is played and the caller is disconnected.
- ***Stop Recording on Silence -*** The selection made here determines whether or not recording of a caller’s answer will be stopped automatically after a short period of silence (4 seconds) is detected. (To change the default setting, contact your Northern Telecom support organization.)

Silence that occurs during recording could mean one of several things:

1 A silence right at the beginning of the caller’s answer (after the record beep is sounded and before voice is detected) could mean that the caller does not intend to answer the question, or that the caller needs a few seconds to think of or compose an answer.

2 A silence that follows voice detection could indicate that the caller has finished answering the question.

3 A silence that follows voice detection could also occur in the middle of an answer if the caller pauses for longer than 4 seconds in order to think more about the answer he or she is giving or to try to remember additional information.

If this field is set to “Yes”, situation 1 would always be considered a “no answer” and the action specified in the *No Answer Handling* field would be taken. If a caller needs time to think before giving an answer, he or she would not have a chance to answer the question unless the *No Answer Handling* field is set to “Repeat”. For situation 2, it would be desirable for recording to stop, however, you cannot be sure that this is the meaning of the silence. For situation 3, it would not be desirable for recording to stop since the caller has not finished his or her answer.

If this field is set to “No”, the system will wait until the amount of time specified in the *Answer Length Limit* field has passed before going to the next field. This might not be so desirable in situation 2 since the caller might have to wait a considerable time (especially if they are using a rotary phone).

Since you cannot anticipate what type of silence the system is detecting, you should try to give the caller as much flexibility as possible. The most flexible solution is to set this field to “No” and inform users (either in the introductory greeting or with an instructional prompt) that they can press number sign (#) once they are finished recording and ready to go to the next question. However, callers without touch-tone phones cannot press number sign and would therefore have to wait until the answer length limit is reached before being presented with the next field.

- ***Field to be Confirmed*** **-** This field is only displayed if the Confirmation Mode for the voice form is configured as “At Each Field” or “Whole Form”. If this field is set to “Yes”, callers will be asked to confirm their answer to this field. If you do not want the field to be confirmed, set this field to “No”.

  If this field is set to “Yes” and if the confirmation mode is “At Each Field”, callers will be asked to confirm their answer for this voice form field immediately after their answer has been recorded.

  If this field is set to “Yes” and if the confirmation mode is “Whole Form”, the caller will be asked to confirm his or her answer at the end of the form (along with any other fields that are marked for confirmation).

- ***Save Response if Hangup*** **-** The selection made in this field determines whether the entire caller response, including all of the answers recorded so far, should be saved (“Yes”) or discarded (“No”) if the caller hangs up while on this field.

  This field should be given some consideration. If you want to be informed of all calls made to the voice form (complete and incomplete), set this field to “Yes” for all fields. If you are not interested in incomplete responses, set this field to “No” for all fields except the last one. If you want to keep incomplete responses only if they contain some useful information, you will have to decide at which point in the form enough useable information has been gathered. In the third case, you would probably set this field to “No” for the fields at the top of the form and then set it to “Yes” for all fields after a particular point (the point at which you believe enough useful information has been gathered). However, be sure that all fields after this point are set to “Yes”, because whether a response is saved or discarded is determined by the setting of the field on which the caller hangs up.

- ***Field Separator Delay*** **-** The setting made in this field affects what happens after a transcriber has heard the answer recorded for this field. Your choices are:

  - ***Stop*** **-** After hearing an answer, playback stops until the transcriber issues the Next Field or Play. This gives the transcriber a chance to transcribe the answer.
  - ***Delay (deciseconds)*** **-** After the answer for this field has been played to the transcriber, there is a delay before the next answer is played during which the transcriber is given the chance to transcribe the answer. The duration of this delay is specified in deciseconds in the adjacent field. The default value is configured in the Add a Voice Form Definition screen.

A decisecond equals one tenth (1/10th) of a second. Therefore, to determine the number of deciseconds required, decide on the required duration of the delay in seconds and then multiply this value by 10. For example, to configure a delay of 15 seconds, enter 150 in this field.

Figure 8-46xxx
The Insert New Field screen for a “No Answer” type field

```
                         Voice Administration
Insert New Field    Form ID: 02      Title: Credit Card Application

Field Type:                         [No-Answer]   Voice-Answer

Field Name:                         Welcome

Field Prompt Recorded (Voice):      No

Action After Field:                 Next Field  [Revert]  Disconnect

*Revert DN:                         0

**System Disconnect Message:        [No]  Yes

Save Response if Hangup:            No   [Yes]
--------------------------------------------------------------------
Move selection bar to a field, then select a softkey>

  Save          Cancel                        Voice
```

```
 * This field is displayed only if Action After Field is Revert.
 **This field is displayed only if Action After Field is Disconnect
```

The following read only-fields are displayed:

- ***Form ID -*** is a read-only field. It serves to remind you of the voice form application you are currently working on.
- ***Title -*** is a read-only field. It serves to remind you of the voice form application you are currently working on.

Fill in the following fields for a “No Answer” field:

- ***Field Type -*** specifies the type of voice form field. The selection you make here will affect which fields are displayed on this screen. Your choices are:
  - ***No Answer -*** This type of field plays a prompt (a greeting or instruction) to the caller. The caller simply listens to the prompt without responding.
  - ***Voice Answer -*** This type of field plays a prompt (a question) to the caller and records an answer. The caller is generally expected to respond. However, this is not necessarily the case, depending how on the *No Answer Handling* field is configured.

- ***Field Name -*** specifies the name of the field. This name should be unique and descriptive enough to identify the function of the field. You may enter a name that is between 1 and 29 characters in length. This is a mandatory field.
- ***Field Prompt Recorded (Voice) -*** indicates whether or not a prompt has been recorded for this field. This prompt is heard by the caller as soon as they reach this field in the voice form. The prompt usually contains information (for No-Answer fields) or a question (for Voice-Answer fields). To record a voice prompt, press the [Voice] softkey while the cursor is positioned on this field. See the section "Recording field prompts and field names" on page 8-162 for more information.

  Note: All field prompts must be recorded before the voice form can be used.

- ***Action After Field -*** This field determines what should happen after the field prompt is played to the caller. Your options are:
  - ***Next Field -*** the prompt recorded for the next field in the voice form is played.
  - ***Revert -*** the caller is transferred to a specified phone number. The DN is defined in the Revert DN field, described below.
  - ***Disconnect -*** the call is disconnected. This action is typically selected for the last field in the voice form that plays the farewell greeting and is used to disconnect the caller from the voice form. (You have the option of playing a system "Goodbye" greeting before disconnecting calls.)
- ***Revert DN -*** This field is only displayed if the *Action After Field* field is set to "Revert". This is the DN to which a caller is transferred after the prompt for this field is played.
- ***System Disconnect Message -*** This field is only displayed if you have chosen "Disconnect" as the Action After Field. It is intended for the last field in the form. When this field is set to "Yes" the caller hears a system recording (*"Goodbye"*) before being disconnected. If you record your own farewell greeting that includes the word "Goodbye", set this field to "No". Otherwise, callers will hear two goodbye messages (the system goodbye and your custom prompt).
- ***Save Response if Hangup*** - The selection made in this field determines whether the entire caller response, including all of the answers recorded so far, should be saved ("Yes") or discarded ("No") if the caller hangs up while on this field.

This field should be given some consideration. If you want to be informed of all calls made to the voice form (complete and incomplete), set this field to “Yes” for all fields. If you are not interested in incomplete responses, set this field to “No” for all fields except the last one. If you want to keep incomplete responses only if they contain some useful information, you will have to decide at which point in the form enough useable information has been gathered. In the third case, you would probably set this field to “No” for the fields at the top of the form and then set it to “Yes” for all fields after a particular point (the point at which you believe enough useful information has been gathered). However, be sure that all fields after this point are set to “Yes”, because whether a response is saved or discarded is determined by the setting of the field on which the caller hangs up.

Recording field prompts and field names

For each voice answer field in the form, you must record both a field name and a field prompt. The field name is played to transcribers during transcription if the Field Name is used as the field separator. It is also played to callers during whole form confirmation. You can record the field name as you define each field. However, you may want to record your field prompts during a separate recording session. See the section “Recording the prompts” in the *Implementation Guide* for a discussion.

Procedure 8-46xxx
Recording field prompts and field names

Starting point The List of Fields screen.

1 If this is the first field to be added to the voice form. go to step 1a. For subsequent fields, go to step 1b.

a. Position the cursor on the <CONFIRMATION> marker (if it is present). If there is no <CONFIRMATION> marker, position the cursor on the <END-OF-FORM> marker.

b. Position the cursor to indicate where the new row should be inserted. The new row will be inserted above the row in which the cursor is placed. If this is the last field in the form, the cursor should be on the <END-OF-FORM> marker.

2 Press the <space bar> to select the row.

3 Press [Insert New Field].

The Insert New Field screen is displayed.

4 To record the field prompt, position the cursor on the *Field Prompt Recorded* field.

a. Press the [Voice] softkey.

You are prompted for an extension number.

b. Enter the extension number of the phone set you are going to use to make the recording and press <Return>.

The phone rings.

c. Pick up the telephone handset.

d. To record the field prompt, press the [Record] softkey. At the sound of the beep, begin speaking into the handset.

When you pressed the [Record] softkey, a new [Stop] softkey appeared in its place. Press the [Stop] softkey to stop recording.

e. To hear the prompt, press the [Play] softkey.

The recording is played over the phone.

If you want to rerecord the prompt, return to step 4d.

f. When you are satisfied with the recording, press either [Disconnect] or [Return] to display the original softkeys.

When you use [Return], the line is not disconnected (unless you hang up the receiver). This means that if you decide to re-record or listen to the re-cording, you do not have to re-enter the telephone extension after pressing the [Voice] softkey. This is recommended if you will be recording a number of prompts.

When you use [Disconnect], the line is disconnected and if you press [Voice] to access the recording softkeys again, you will have to re-enter the telephone extension. Use this softkey when you have recorded the last prompt for this session.

5 To record the field name, position the cursor on the *Field Name Recorded* field. Follow steps 4a to 4f.

6 Press [Save] to save the recordings.

7 To save the entire field definition, go to step 7a. If you do not want to save the field definition, go to step 7b.

a. Press [Save].

If all of the mandatory fields have been filled in, the field definition is saved. If something is missing from your definition, you will be informed and the field will not be saved until all of the required information has been entered.

Note: *When you return to the Add a Voice Form Definition, be sure to use the [Save] softkey that is displayed. If you do not save the voice form, the fields you have just created will not be saved.*

b. Press [Cancel].

Any changes that have been made are discarded. The List of Fields screen is displayed.

Saving the voice form definition

Follow Procedure 8-47 when you have defined all of your fields. If the confirmation mode is “Whole Form”, make sure the <CONFIRMATION> marker is positioned correctly.

Procedure 8-47xxx
Saving the entire voice form definition

Starting point The Insert New Field screen.

1 Press [Close Fields].

The Add a Voice Form Definition screen is displayed.

2 Press [Save] to save the entire voice form definition, including fields.

The Voice Forms screen is displayed.

Making voice forms accessible

You can make a voice form accessible to callers in one of three ways:

Direct Access

Callers access the form by dialing a special DN that connects them directly to the voice form.

Define a DN for the voice form in the VSDN table. The type of service is VF (Voice Form). See the section “The Voice Services-DN Table” on page 8-50.

If there are no available DNs on the DMS/SL-100, configure one for each voice form that will be directly dialable. See the section “Configuring voice services” on page 8-29.

Indirect access through a voice menu

Callers dial the DN of a voice menu. The voice form is presented as one of the menu choices and is accessed when the caller presses the appropriate key.

If the voice menu service already exists, modify it so that the appropriate voice form ID is associated with a particular key or create a new voice menu application if necessary. If you create a new voice menu, define a DN for the voice menu in the VSDN table. See the following sections: “Adding a voice menu” on page 8-103 and “The Voice Services-DN Table” on page 8-50.

If there are no available DNs on the DMS/SL-100, configure one for each voice form that will be directly dialable. See the section “Configuring voice services on the DMS/SL-100” on page 8-29.

Indirect access through a time-of-day controller

Callers dial the DN of the active time-of-day controller and are routed to the voice form depending on the day and time of day.

Modify the time-of-day controller definition so that the appropriate voice form ID is associated with business hours, off-hours, and/or holidays. If you need to create a new time-of-day controller, define its DN in the VSDN table. See the following sections: “Adding a Time-of-Day Controller” on page 8-90 and “The Voice Services-DN Table” on page 8-50.

If there are no available DNs on the DMS/SL-100, configure one for each voice form that will be directly dialable. See the section “Configuring voice services on the DMS/SL-100” on page 8-29.

Defining a VSDN for the transcription service

After you have created your first voice form application, define a DN for the transcription service in the VSDN table. Transcribers have to access the transcription service (much like you would access voice messaging) in order to transcribe a voice form.

You can have one generic transcription service which is used by all transcribers to log on to all voice forms. Follow Procedure 8-48.

Procedure 8-48xxx
Defining the transcription service VSDN

Starting point The Main Menu.

1 Select Voice Administration.

2 Select Voice System Configuration/Voice Menu Applications Administration.

3 Select Voice Services-DN Table.

4 Press the [Add] softkey to define a DN for the transcription service.

The Add DN Information screen is displayed.

5 Enter the Access DN for the service.

6 In the *Service* field, enter TR (for transcription).

You are prompted for the voice form ID.

7 Leave the *ID* field blank.

8 Save the DN information.

Because you left the Voice Form ID blank in the above procedure, transcribers are required to provide the voice form ID when they log on.

If you want to create a special transcription DN for each voice form application, enter the voice form ID in the *ID* field instead of leaving it blank (see step 7 in the above procedure). Using this method, transcribers do not have to specify the voice form ID when they log on. However, they will have to remember a different transcription service access DN for each voice form. Also keep in mind that you will need a DN on the DMS/SL-100 for each VSDN you add in DMS VoiceMail.

If there are no available DNs on the DMS/SL-100, configure one for each voice form that will be directly dialable. See the section “Configuring voice services on the DMS/SL-100” on page 8-29.

For more information about adding DNs, see the section “The Voice Services-DN Table” on page 8-50. Also see the *Meridian Voice Forms Transcriber User Guide* for a description of how transcribers log on to voice forms.

Testing a voice form

When you have finished configuring a voice form and have defined a DN for the form and the transcription service, test it before making it available. Test the caller interface and the transcriber interface. If you have to make any modifications to the voice form based on your testing, be sure to test the form again. (See the following section, “Modifying a voice form definition”.) Continue with this process until you are satisfied with the way in which the voice form and the transcription service operate.

See the *Meridian Voice Forms Implementation Guide* for more information about testing. This guide also includes two surveys: one for callers and one for transcribers. Give these surveys to the people who test your form and use the feedback to fine tune your application.

Modifying or viewing a voice form definition

After testing your voice form, you will probably discover that modifications are necessary to make the voice form work properly or more smoothly. Depending on the application, you may also need to modify a voice form on a regular basis in order to keep it up-to-date.

When you select the [View/Modify] softkey from the Voice Forms screen, a second layer of softkeys is displayed (as shown in Procedure 8-51). There are two modify softkeys: [Modify In-Service] and [Modify Out-of-Service].

Modifying a voice form while it is in-service

When you choose to keep a form in-service when modifying it, incoming calls to the form and existing caller responses are not affected. However, because the form is still functional, you can only make simple changes to it that do not affect its structure. You can modify all of the fields in the Modify Field screen except *Field Type*. You can also re-record voice prompts without affecting incoming calls. You cannot, however, make changes that will alter the structure of the voice form application.

While the form is in-service, you cannot:

- insert, delete, or move the fields within the selected form
- change the Field Type (from “No Answer” to “Voice Answer” or vice-versa).

If you need to make any of these changes, you will have to use the [Modify Out-Of-Service] softkey.

Modifying a voice form while it is out-of-service

While a voice form is out-of-service you can make any changes you wish. While the form is out-of-service, callers that try to connect to the form are told that the form is not available at the time and to try calling again at a later time. If you take the form out-of-service while a call is in progress, the caller’s response will be discarded when he or she hangs up.

When you press the [Modify Out-of-Service] softkey, the system checks for any untranscribed responses. If there are any outstanding caller responses, a message is displayed on the command line. A new layer of softkeys is also presented, allowing you to [Delete All Responses] or [Cancel] the current command. To proceed with the modification you must transcribe all existing responses first or press the [Delete All Responses] softkey. (This is not recommended unless you are sure you don’t need the responses.)

If the voice form generates a lot of calls, the best way to put the form out of service (without dropping calls or losing responses) is to build an announcement that informs callers that the voice form is currently being serviced. To do this:

Procedure 8-49xxx
Taking a voice form out of service

1. Create an announcement. You can make a single “generic” announcement which can be used whenever you need to take a form out of service. See the section “Adding an Announcement” earlier in this chapter.
2. Reassign the VSDN for the voice form to the announcement service. See the section “The Voice Services-DN Table” earlier in this chapter.
3. Transcribe any existing responses. See the *Meridian Voice Forms Transcriber User Guide*. (You can ask the transcriber to do this step.)
4. Modify the voice form and save the changes. See Procedure 8-51.
5. Reassign the announcement VSDN back to the voice form.

Use Procedure 8-50 if you do not need to make changes to the voice form definition but need to view its configuration. If, while viewing the definition, you discover that you need to make a change, you will have to get out of the View a Voice Form Definition screen and access the Modify a Voice Form Definition screen. Use Procedure 8-51 if you need to modify the voice form definition.

Procedure 8-50xxx
Viewing a voice form definition

Starting point The Voice Forms screen.

1 Position the cursor on the voice form definition that you want to view and press the <spacebar> to select it.

2 Press [View/Modify].

A new set of softkeys is displayed.

```
Select a softkey >
            Cancel      View Only     Modify        Modify
                                    In-Service   Out-of-Service
```

3 Press the [View Only] softkey.

The View/Modify a Voice Form Definition screen is displayed. (It is identical to the Add a Voice Form Definition screen.)

4 Press the [Open Fields] softkey if you want to view any of the field definitions.

The List of Fields screen is displayed (Figure 8-44).

5 Move the cursor to the field definition that you want to view and press the <spacebar> to select it.

6 Press the [View Field] softkey.

The View Field screen is displayed.

Procedure 8-51xxx
Modifying a voice form definition

Starting point The Voice Forms screen.

1 Position the cursor on the voice form definition that you want to modify and press the <spacebar> to select it.

2 Press [View/Modify].

The following softkeys are displayed:

```
Select a softkey >
            Cancel      View Only     Modify        Modify
                                    In-Service   Out-of-Service
```

3 Press the [Modify In-Service] or [Modify Out-of-Service] softkey.

Note: *Be sure you understand the difference between these two methods of modifying a voice form before proceeding. See page 8-166 for details.*

If you selected [Modify In-Service] the View/Modify a Voice Form Definition screen is displayed. (It is identical to the Add a Voice Form Definition screen.)

If you selected [Modify Out-of-Service] and there are outstanding untranscribed responses, a subset of softkeys is displayed.

```
Caller responses exist for this voice form. Select a softkey. >
|          |  Cancel  | Delete All |          |          |
|          |          | Responses  |          |          |
```

Important: *Do not press the [Delete All Responses] softkey unless you are sure you do not need the responses. Use [cancel] and exit the form to transcribe responses. See page 8-167 for more information.*

Press the [Cancel] softkey and exit the voice form. If it is important that you do not lose any calls or responses while servicing the form, see Procedure 8-49. Transcribe existing responses and log back onto the form (return to step 1).

If you are sure that the responses are not needed (for example, they are invalid because the form is out-of-date), press the [Delete All Responses] softkey.

4 Press the [Open/Modify Fields] softkey if you want to modify any of the field definitions.

The List of Fields screen is displayed (Figure 8-44).

5 Move the cursor to the field definition that you want to view and press the <spacebar> to select it.

6 Press the [View/Modify Field] softkey.

The View/Modify Field screen is displayed.

Modifying an existing field

Once you have created the fields that make up your voice form applications, they can be modified at any time. The View/Modify Field screen is identical to the Insert New Field screen (see page 8-155). For a description of the fields in this screen, see the preceding section, “Inserting a new field”.

> ***Note:*** If you want to change the Field Type (from “No Answer” to “Voice Answer” for example), you will have to modify the form while it is out-of-service.

Procedure 8-52xxx
Modifying an existing field

Starting point The List of Fields screen.

1 Position the cursor on the field that you want to modify and press the <spacebar> to select it.

2 Press [View/Modify Field].

The View/Modify Field screen is displayed.

3 Make the necessary changes, including re-recording the field prompt and field name if required.

4 To save the field definition, go to step 4a. If you do not want to save the field definition, go to step 4b.

a. Press [Save].

If all of the mandatory fields have been filled in, the field definition is saved. If something is missing from your definition, you will be informed and the field will not be saved until all of the required information has been entered.

b. Press [Cancel].

Any changes that have been made are discarded. The List of Fields screen is displayed.

Moving a field

It may be necessary to move some fields around if you decide that the application would flow more smoothly if you changed the order of voice form fields.

The <CONFIRMATION> marker can be moved to a different location in the field list but cannot be modified or deleted. This enables you to place one or two additional questions after the confirmation field. These questions could, for example, inquire as to the caller’s opinion about the current level of service he or she is presently receiving when dealing with your organization. It is also recommended that you play a thank you or farewell prompt to the caller after confirmation, before disconnecting the call.

> ***Note:*** The form must be out-of-service in order to move a field.

Procedure 8-53xxx
Moving a field

Starting point The List of Fields screen.

1 Position the cursor on the field you want to move and press the <spacebar> to select it.

2 Press the [Move Field] softkey.

The following softkeys are displayed:

```
Caller responses exist for this voice form. Select a softkey. >
              Cancel                                           Done
```

3 Press the <spacebar> to deselect the field you want to move.

This allows you to move the cursor to indicate the new position.

4 Move the cursor to indicate where the row should be inserted. (When positioning the cursor remember that the row will be moved to the row above the cursor location.)

5 Press the <spacebar> to indicate the new location.

6 Press [Done] to continue or [Cancel] to cancel the operation.

The field is moved to the specified location.

The Move Field softkeys are removed from the screen and the original List of Fields softkeys are displayed.

7 To move another field, go to step 1.

Deleting a field

Once you have created your voice form application, you may discover that certain fields are not very effective or that they have become obsolete with time. The [Delete Field] softkey on the List of Fields screen allows you to remove fields that are no longer needed.

> ***Note:*** The form must be out-of-service in order to delete a field.

Procedure 8-54xxx
Deleting a field

Starting point The List of Fields screen.

1 Position the cursor on the field you want to delete and press the <spacebar> to select it.

2 Press the [Delete Field] softkey.

The Delete Field screen is displayed. This screen is identical to the Insert New Field screen except that all of the fields are read-only. This allows you to view the field definition and verify that this is the field you want to delete.

3 To delete the field, go to step 3a. To cancel the operation, go to step 3b.

a. Press [OK to Delete].

The field is deleted and the List of Fields screen is displayed.

b. Press [Cancel].

The field is not deleted. The List of Fields screen is displayed.

Inserting a new field

If you need to insert a new field in an existing voice form, follow Procedure 8-55. Because this affects the structure of the voice form, you will have to modify the form while it is out-of-service.

Procedure 8-55xxx
Inserting a new field

Starting point The List of Fields screen.

1 If this is the first field to be added to the voice form. go to step 1a. For subsequent fields, go to step 1b.

a. Position the cursor on the <CONFIRMATION> marker (if it is present). If there is no <CONFIRMATION> marker, position the cursor on the <END-OF-FORM> marker.

b. Position the cursor to indicate where the new row should be inserted. The new row will be inserted above the row in which the cursor is placed. If this is the last field in the form, the cursor should be on the <END-OF-FORM> marker.

2 Press the <space bar> to select the row.

3 Press [Insert New Field].

The Insert New Field screen is displayed (see page 8-155).

4 Select the Field Type.

Your selection will affect the screen display. See Figure 8-45 for “Voice Answer” fields. See Figure 8-46 for “No Answer” fields.

5 Fill in the required information (according to the New Field worksheets that are provided in the Implementation Guide, if you used them). Record a field prompt and field name. The fields in this screen are described on the following pages.

6 To save the field definition, go to step 6a. If you do not want to save the field definition, go to step 6b.

a. Press [Save].

If all of the mandatory fields have been filled in, the field definition is saved. If something is missing from your definition, you will be informed and the field will not be saved until all of the required information has been entered.

Note: *When you return to the Add a Voice Form Definition, be sure to use the [Save] softkey that is displayed. If you do not save the voice form, the fields you have just created will not be saved.*

b. Press [Cancel].

Any changes that have been made are discarded. The List of Fields screen is displayed.

Copying a voice form definition

If the voice form application you are about to create is similar to an existing one you may want to copy the existing definition, and then modify only those fields that need to be changed. Depending on how long the voice form is, this method can save you a lot of time.

When you copy a voice form, all fields will remain the same with the exception of:

- the voice form ID
- the new response notification DN
- the special response notification DN

These fields are intentionally left blank when you a copy a form because the values in these fields must be unique across all voice forms and voice menu applications. You are required to enter new values.

When you copy from an existing definition, you will be working with the Copy a Voice Form Definition screen. The fields in this screen are identical to those in the Add a Voice Form Definition screen (see Figure 8-42).

Procedure 8-56xxx
Copying a voice form definition

Starting point The Voice Forms screen.

1 Position the cursor on the voice form definition that you want to copy and press the <spacebar> to select it.

2 Press the [Add/Copy] softkey.

3 A new set of softkeys is displayed.

```
Select a softkey. >
            Cancel      Add New     Copy Form
                         Form
```

4 Select [Copy Form].

The Copy a Voice Form Definition screen is displayed. It is identical to the Add a Voice Form Definition screen shown in Figure 8-42 except that the following fields are blank and must be redefined: Voice Form ID, New Responses Notification DN, Special Responses Notification DN.

5 Assign a new voice form ID and modify the necessary fields. These fields are described in the section, “Adding a voice form definition”.

6 Press [Open/Modify Fields].

The List of Fields screen is displayed (Figure 8-44). This screen lists all of the fields that have been defined for this voice form application.

7 Move the cursor to one of the fields that you want to modify.

8 Press [View/Modify Field] to modify the field definition.

The View/Modify Field screen is displayed (it is identical to the Insert New Field screen). Modify any fields and recordings that need to be changed to suit the new application. These fields are described beginning on page 8-155.

9 Press [Save] to save the field definition.

The List of Fields screen is displayed. To modify another field, go to step 7. To exit this screen go to step 10.

10 Press [Close Fields].

The Copy a Voice Form Definition screen is displayed.

11 Press [Save] to save the voice form definition.

The Voice Forms screen is displayed.

Deleting a voice form definition

Selecting the [Delete] softkey from the Voice Forms screen allows you to remove an existing voice form definition from the system. The Delete a Voice Form Definition screen is displayed so that you can view the voice form definition and confirm that it is really the one you want to delete. The fields in this screen are read-only.

You will not be able to delete a voice form definition if there are any outstanding untranscribed responses unless you first transcribe or delete all of the untranscribed responses. When you select the [Delete] softkey a menu prompt appears on the screen's command line to advise you if there are any outstanding caller responses. A new layer of softkeys is also presented, allowing you to [Delete All Responses] or [Cancel] the current command. If you choose to delete all responses, any calls that are currently in progress will be discarded when the caller(s) hang up and any future calls to the voice form application will result in an appropriate 'Out of Service' recording. This remains in effect until the VSDN for the voice form has been removed or the form has been removed from the menu service or time-of-day controller through which it is accessed. Refer to the procedure on page 8-167.

Procedure 8-57xxx
Deleting a voice form definition

Starting point The Voice Forms screen.

1 Position the cursor on the voice form definition that you want to delete and press the <spacebar> to select it.

2 Press the [Delete] softkey.

If there are no outstanding caller responses, the Delete a Voice Form Definition screen is displayed. This allows you to view the configuration and ensure that this is the voice form that you want to delete. Go to step 3.

If there are any caller responses that have not been transcribed, the following softkeys are displayed.

```
Caller responses exist for this voice form. Select a softkey. >
              Cancel      Delete All
                          Responses
```

Important: *Do not press the [Delete All Responses] softkey unless you are sure you do not need the responses. Use [cancel] and exit the form to transcribe responses.*

Press the [Cancel] softkey and exit the voice form. If it is important that you do not lose any calls or responses while servicing the form, see Procedure 8-49. Transcribe existing responses and log back onto the form (return to step 1).

If you are sure that the responses are not needed (for example, they are invalid because the form is out-of-date), press the [Delete All Responses] softkey.

3 To delete the voice form go to step 3a. To cancel go to step 3b.

a. Press [OK to Delete].

The voice form is deleted. The Voice Forms screen is displayed.

b. Press [Cancel].

The voice form is not deleted. The Voice Forms screen is displayed.

4 If this form is directly accessed, remove its VSDN from the VSDN table. If the form is accessed through a voice menu or time-of-day controller, delete the voice form from the appropriate voice menu or time-of-day controller definition.

Hardware Administration

Hardware Administration allows you to view the contents of the hardware database in your DMS VoiceMail system. The hardware database is a system utility which maintains a current listing and description of all nodes, cards, T1 channels, and DSP ports in your system. If you need to modify the hardware database, you (or a representative from your support organization) must use the Modify Hardware utility. This utility is documented in *Appendix A: System Administration Tools*.

CAUTION
Overnight system audits
You should not leave the administrative console in any Hardware Administration menu overnight or important system audits may fail due to lack of available memory.

The Hardware Administration menu

From the Hardware Administration menu (Figure 9-1) you can choose to view your system's node configuration, data port configuration and T1 channel configuration. You can also print this information using one of the Print options in the Hardware Administration menu.

Figure 9-1xxx
The Hardware Administration menu

```
                         Hardware Administration

                  1  Node Configuration
                  2  T1 Link Configuration
                  3  Data Port Configuration
                  4  Print All Node Information
                  5  Print All Data Port Information

 ------------------------------------------------------------------
 Select an item >

   Exit
```

Procedure 9-1xxx
Using the Hardware Administration menu

Starting point : The Main Menu, item <4> selected.

The Hardware Administration menu appears (Figure 9-1).

1 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears. See the following sections for details:

<1> "Node Configuration"
<2> "T1 Link Configuration"
<3> "Data Port Configuration"
<4> "Printing node or data port information"
<5> "Printing all data port information".

2 Use [Exit] to return to the Main Menu.

Node Configuration

The Node Configuration screen (Figure 9-2) is a summary listing of the cards found on all nodes in your system.

Figure 9-2xxx
Node Configuration screen

```
                      Hardware Administration

  Node Configuration


  Node Card_1    Card_2    Card_3

  1    Empty     SBC       Bus
  2    Bus       Empty     SBC
  3    VP12      VP12      SBC
  4    VP12      VP12      SBC
  5    VP12      VP12      SBC
  6    VP12      VP12      SBC
  ..   ...       ...       ...
  13   T1        Empty     SBC
  14   SBC       Empty     T1

Move the cursor to the node number and press the space bar to select.

    Exit          View
```

Note: The figures in this section do not necessarily represent actual hardware configurations. They are illustrations only.

The following fields are displayed:

- ***Node -*** The node number.
- ***Card -*** The types of cards found on the specified node. The following abbreviations identify the following cards:
  - ***SBC*** - single board computer (also known as the 68k card)
  - ***Bus*** - high-speed bus
  - ***VP12*** - 12-channel voice processor
  - ***T1*** - T1 link

Procedure 9-2xxx
Viewing node configurations

Starting point : The Hardware Administration screen, item <1> selected.

The Node Configuration screen appears (Figure 9-2).

1 Move the cursor to the node you want to view and press <Space Bar>.

Your selection is highlighted.

2 Choose step 2a to view the configuration information of the node or 2b to return to the Hardware Administration menu.

a. Use [View].

The View Node screen appears; see the next section, “Viewing nodes”.

b. Use [Exit].

The Hardware Administration screen is redisplayed.

Viewing nodes

The View Node screen (Figure 9-3) displays the cards and ports (and their attributes) that are installed on the node you selected in the Node Configuration screen.

Figure 9-3xxx
View Node screen

```
                              Hardware Administration

View Node

Location  Card_Type  Port_Type  Attributes

1-1-*     Empty

1-2-*     SBC
1-2-1                Data:      [Terminal]Printer NWModem MMLink AML/CSL  SMDI  PMS AdminPlus
1-2-2                Data:       Terminal Printer NWModem MMLINK AML/CSL [SMDI] PMS AdminPlus
1-2-3                Data:       Terminal Printer NWModem MMLINK AML/CSL [SMDI] PMS AdminPlus
1-2-4                Device:    [Disk] Tape
1-2-5                Device:    [Disk] Tape

1-3-*     Bus

                                                                          MORE BELOW

  Exit
```

```
                              Hardware Administration            MORE ABOVE

View Node

Location  Card_Type  Port_Type  Attributes

14-1-*    SBC
14-1-1               Data       Terminal Printer NWModem  MMLink AML/CSL [SMDI] PMS AdminPlus
14-1-2               Data       Terminal Printer NWModem  MMLink AML/CSL [SMDI] PMS AdminPlus
14-1-3               Data       Terminal Printer NWModem  MMLink AML/CSL [SMDI] PMS AdminPlus
14-1-4               Data       Terminal Printer NWModem  MMLink AML/CSL [SMDI] PMS AdminPlus

14-2-*    Empty

14-3-*    T1
14-3-1                Link
14-3-2                Link
14-4-3                Link
14-4-4                Link
```

Note: The figures in this section do not necessarily represent an actual hardware configuration. They are presented for illustration purposes only. If the node you are viewing is a system node you may have the following types of cards installed: SBC or Bus. A voice node would have the following types of cards installed: SBC and VP12. A TIFN node would have the T1 card and SBC card installed.

The screen displays the following read-only information about each card on the node:

- ***Location*** - The physical location of the card in the DMS V oiceMail system. The location is identified by the node-card-port number.
- ***Card Type*** - The function of the card; see “Node Configuration” for a description of the abbreviations used in this field.
- ***Port Type*** - The type of port. “Data” indicates a serial data communications port. “Device” indicates a mass storage device or tape drive. “Voice” indicates a voice processor port. “Link” indicates a T1 link.
- ***Attributes*** *(for ports with port type = Data)*
  - ***Terminal:*** Indicates a connection to an administration terminal or a personal computer.
  - ***MMLink:*** Meridian ACCESS Link. This is the communications channel for Meridian ACCESS. This is an optional feature that is available on CPE systems only.
  - ***AML/CSL or Meridian Link:*** Not applicable.
  - ***SMDI:*** Simplified Message Desk Interface. This is the communications channel between DMS VoiceMail and the local switch for Call Progress information.
  - ***Printer:*** Printer serial connection.
  - ***NWModem:*** Connection to a modem used for networking calls.
  - ***PMS***: Not applicable.
  - ***AdminPlus:*** Indicates a connection to a PC running AdminPlus software.
- ***Attributes*** *(for ports with port type = Device)*
  - ***Disk:*** Mass storage subsystem (hard disk)
  - ***Tape:*** Cartridge tape subsystem

T1 Link Configuration

The T1 Link Configuration screen lists the T1 channels in the DMS VoiceMail system.

Figure 9-4xxx
T1 Link Configuration screen

```
                         Hardware Administration

T1 Link Configuration


               Primary Connection  Secondary Connection  T1 Clock Reference
T1 Link ID     (Node-Card-Span)     (Node-Card-Span)         Candidacy

     A            13-1-1              14-1-1                   Y
     B            13-1-2              14-1-2
     C            13-1-3              14-1-2
     D            13-1-4              14-1-4
     E            15-1-1              16-1-1                   Y
     F            15-1-2              16-1-2
     G            15-1-3              16-1-3
     H            15-1-4              16-1-4                   Y

Move the cursor to the item and press the space bar to select.
    Exit      Modify T1 Chnl    Modify T1
              Configuration    Link Setup
```

Note: The figures in this section do not necessarily represent actual hardware configurations. They are illustrations only.

The following fields are displayed on this screen:

- ***T1 Link ID -*** A unique identifier for the T1 link. Each link actually consists of two connections, a primary and secondary connection, to provide redundancy.
- ***Primary Connection -*** The location (node-card-span) of the primary connection.
- ***Secondary Connection -*** The location (node-card-span) of the secondary connection.
- ***T1 Clock Reference Candidacy -*** This field shows whether or not the link has been configured as a candidate for clock referencing. Use the [Modify T1 Link Setup] softkey to nominate a link or to disqualify a current candidate. See the section “Modifying the T1 link setup” for more information about clock referencing.

Procedure 9-3xxx
Viewing T1 link configurations

Starting point : The Hardware Administration screen, item <2> selected.

The T1 Link Configuration screen appears (Figure 9-4).

1 Move the cursor to the T1 link you want to modify and press <Space Bar>.

Your selection is highlighted.

2 Choose step 2a to modify the T1 channel configuration information of the link. Choose step 2b to modify the T1 link setup information. Choose step 2c to return to the Hardware Administration menu.

a. Use [Modify T1 Chnl Configuration].

The Modify T1 Channel Configuration screen is displayed. See the next section, "Modifying T1 channels".

b. Use [Modify T1 Link Setup].

The T1 Link Setup screen is displayed. See the section "Modifying the T1 link setup".

c. Use [Exit].

The Hardware Administration screen is redisplayed.

Modifying T1 channels

The Modify T1 Channel screen (Figure 9-5) displays the T1 Channel configuration for the link you select.

Figure 9-5xxx
The Modify T1 Channel screen

```
                         Hardware Administration

Modify T1 Channel Configuration for T1 Link ID A

Channel   Routing     Login  Logout    Agent    Not-ready
Number    Address     Code    Code     ID Code  Deactivation Code


    1     0 -1234     1234   1234      1234
    2     0 -5432     2222   3333      4444
    3     0 -0
    4     0 -0
    5     0 -0
    6     0 -0
    7     0 -0
    8     0 -0
    9     0 -0
   10     0 -0
   11     0 -0
   12     0 -0
   13     0 -0
   14     0 -0
   15     0 -0
   16     0 -0
   17     0 -0
   18     0 -0
   19     0 -0
   20     0 -0
   21     0 -0
   22     0 -0
   23     0 -0
   24     0 -0

   Save        cancel
```

Note: The figures in this section do not necessarily represent an actual hardware configuration. They are presented for illustration purposes only.

The following fields are displayed on this screen:

- ***Channel Number -*** The number of the T1 channel.
- ***Routing Address -*** The location of the corresponding agent in the switch. This is the Message Desk Number and is represented in the format xx-yyyy, where xx is the message desk number and yyyy is the terminal number.

- ***Login Code -*** The channel access code for logging in to the UCD group. This field should be blank if the SMDI_AUTOLOG option is set to "Y" (yes) on the switch. When this field is left blank, DMS VoiceMail inserts a default login code.

  If SMDI_AUTOLOG is set to "N" on the switch, make sure that the code displayed here matches the code configured on the switch. This code can be obtained from your DMS administrator.

- ***Logout Code -*** The channel access code for logging out of the UCD group. This field should be blank if the SMDI_AUTOLOG option is set to "Y" (yes) on the switch. When this field is left blank, DMS VoiceMail inserts a default login code.

  If SMDI_AUTOLOG is set to "N" on the switch, make sure that the code displayed here matches the code configured on the switch. This code can be obtained from your DMS administrator.

- ***Agent ID Code -*** This code corresponds to the line number (SMDI_LINE_NO) of the agent as configured on the DMS/SL-100. The LINE_NO can either be configured through the servord (so) or through Table IBNFEAT by specifying the SMDI option.
- ***Not-ready Deactivation Code -*** This field is not applicable to DMS UCD environments and should be left blank. It is used in DMS ACD environments for putting the channel to the ACD queue after the channel has logged into the ACD group.

See the *Translations Guide* (NTP 297-7001-310) for more information about these codes.

Modifying the T1 link setup

The T1 Link Setup screen (Figure 9-6) is used to modify the T1 clock reference candidacy of a T1 link. You may nominate one or more links to serve as the clock reference for the SPM (MSM). An external device in the network (such as the DMS-100, for example) serves as the reference provider.

The actual link that is used as the reference is defined in the T1 Link Status screen (see the “System Status and Maintenance” chapter). If any problems occur on the link that is the current clock reference, or if certain maintenance procedures are being carried out on the link or the card, the system will automatically select one of the other nominated links as the new reference and generate a SEER to indicate that a link has been activated as the reference provider. The following situations will cause the system to select another reference.

- a red alarm is detected
- a yellow alarm is detected
- there is a hardware fault
- the T1 card on which the link resides is disabled
- the TIFN is disabled
- the switch T1 link command is issued
- the T1 link that is the clock reference is disabled

In order to nominate a T1 link for clock reference candidacy, you must first take both the primary and secondary spans associated with the T1 link out-of-service. T1 links are enabled and disabled in the T1 Link Status screen (described in the “System Status and Maintenance” chapter).

Figure 9-6xxx
The Modify T1 Link Setup screen

```
                     Hardware Administration
T1 Link Setup for Link ID A

T1 Clock Reference Candidacy:   [No]   Yes

Select a softkey >

 Save *     Cancel
```

*   If you have not disabled the primary and secondary spans, only the [Exit] softkey is displayed and the screen is read-only.

The following field is displayed on this screen:

- ***T1 Clock Reference Candidacy -*** "Y es" indicates that the selected T1 link is nominated as a clock reference candidate. "No" indicates that the link has not been nominated.

Procedure 9-4xxx
Nominating/disqualifying a T1 link as a clock reference candidate

Starting point : The Main Menu.

1 Select System Status and Maintenance.

2 Select T1 Link Status.

3 Press [Disable T1].

You are prompted for the number of the link you want to disable.

4 Enter the number of the link you want to disable followed by <Return>.

To disable another link, repeat steps 3 and 4.

5 Press [Exit].

The System Status and Maintenance menu is displayed.

6 Press [Exit].

The Main Menu is displayed.

7 Select Hardware Administration.

8 Select T1 Link Configuration.

9 Move the cursor to the T1 link you want to nominate/disqualify and press <Space Bar> to select it.

Your selection is highlighted.

10 Press [Modify T1 Link Setup].

The T1 Link Setup screen is displayed.

11 Select “Yes” to nominate a link or “No” to disqualify a current candidate.

12 Press [Save].

The select link is nominated/disqualified and the T1 Link Configuration screen is displayed.

13 Return to the T1 Link Status screen in System Status and Maintenance and re-enable the link(s).

14 If necessary, activate one of the candidates as the clock reference using [Change T1 Clocking Mode] in the T1 Link Status screen. See the section “T1 Link Status” in the “System Status and Maintenance” chapter for more information.

Data Port Configuration

The Data Port Configuration screen (Figure 9-7) summarizes the data ports on all nodes in your system. From this screen you can select a data port and view the configuration. The only data port that can be modified from Hardware Administration is NWModem (applicable only if Networking is installed). The abbreviations used in this screen are described under “Node Configuration” earlier in this chapter.

Figure 9-7xxx
Data Port Configuration screen

```
                        Hardware Administration

Data Port Configuration

Port Location       Description           Device Type      Status

  1-3-1             Node 1  SBC Port 1    Terminal         InService
  1-3-2             Node 1  SBC Port 2    MMLink           InService
  1-3-1             Node 1  SBC Port 3    Terminal         InService
  1-3-2             Node 1  SBC Port 4    Printer          OutofService
  2-3-1             Node 2  SBC Port 1    Printer          InService
  2-3-2             Node 2  SBC Port 2    Printer          OutofService
  2-3-3             Node 2  SBC Port 3    Terminal         InService
  2-3-4             Node 2  SBC Port 4    Terminal         InService
 12-1-2             Node 12 SBC Port 2    NWModem          InService
 13-1-1             Node 13 SBC Port 1    SMDI             InService

Move the cursor to the data port location and press space bar to select.

    Exit       View/Modify
```

The Data Port Configuration screen displays the following information:

- ***Port Location*** - The port’s physical location (node-card-port) in the system.
- ***Description*** - The node and card type on which the port resides.
- ***Device Type*** - The function of the port. SBC port 1 must be set to Terminal. SBC port 1 on node 13 should be set to SMDI.
- ***Status*** - The current operational state of the port.

Procedure 9-5xxx
Viewing data ports

Starting point : The Hardware Administration screen, <3> entered.

1 The Data Port Configuration screen appears (Figure 9-7).

2 Move the cursor to port to be viewed and press <Space Bar>.

Your selection is highlighted.

3 Choose step 3a to view or modify the configuration information, or 3b to return to the Hardware Administration screen.

a. Use [View/Modify].

The View Data Port screen is displayed for the selected device (if NWModem was selected, the Modify Data Port screen is displayed). See the next section for details.

b. Use [Exit].

The Hardware Administration screen appears.

View Data Port

The following sections describe the different Data Port screens which can be displayed. The screen that is displayed is determined by the data port that is selected in the Data Port Configuration screen when you press [View/Modify]. All screens, except the NWModem screen, are read-only.

Terminal Data Ports

The View Data Port screen for terminals (Figure 9-8) allows you to view information about the terminal connected to the selected port.

Figure 9-8xxx
View Data Port screen (Terminal)

```
                        Hardware Administration

View Data Port

Data Port Location:            1-1-1

Device Type:                   Terminal

Device Name:                   CON181

Baud Rate:                     1200 [2400] 4800 9600

Parity:                        Even Odd  [None]

Number of Windows:             6

Window Width:                  80

Window Height:                 24
______________________________________________________________________

Select a softkey >

  Exit
```

The following read-only fields are displayed in the screen:

- ***Data Port Location*** - The physical location of the port. A terminal must be located on node 1, SBC port 1. Other terminals can be in the system on other data ports.
- ***Device Type*** - “Terminal” will be displayed.
- ***Device Name*** - The name that identifies the terminal.
- ***Baud Rate*** - This setting depends on the current set-up of the terminal on the port.
- ***Parity*** - The method by which data is communicated. This can be set to “Even”, “Odd”, or “None”, depending on the current set-up of the terminal connected to the port. It is usually set to “None”.

- ***Number of Windows*** - This field specifies the number of windows that can be used simultaneously. This will be "6" for the System Administration terminal.
- ***Window Width*** - This field specifies the window width used.
- ***Window Height*** - This field specifies the window height used.

Procedure 9-6xxx
Viewing the terminal data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the terminal data port you want to view.

2 Press the <Space Bar> to select it.

3 Press [View/Modify].

The View Data Port screen (for the selected Terminal) is displayed.

4 Press [Exit] to return to the Data Port Configuration screen.

Printer Data Ports

The View Data Port screen for printers (Figure 9-9) allows you to view the baud rate and parity of the printer that is connected to the selected port.

> ***Note 1:*** A secondary printer can be attached directly to the administration terminal. It does not require a separate data port.
>
> ***Note 2:*** SEERs and Operational Measurement reports must be directed to a particular printer. The printer is specified in the General Options screen (see the "General Administration" chapter.)

Figure 9-9xxx
View Data Port screen (Printer)

```
                    Hardware Administration

View Data Port

Data Port Location:          1-3-4

Device Type:                 Printer

Device Name:                 PRT134

Baud Rate:                   1200 2400 4800 [9600]

Parity:                      Even Odd [None]

__________________________________________________________

  Exit
```

The following read-only fields are displayed in the screen:

- ***Data Port Location*** - The physical location (node-card-port) in the system.
- ***Device Type*** - The function of the port. This will be set to "Printer".
- ***Device Name*** - The name of the device.
- ***Baud Rate*** - The setting will depend on the current set-up of the printer connected to the port.
- ***Parity*** - The setting will depend on the current set-up of the printer connected to the port.

Procedure 9-7xxx
Viewing the printer data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the printer data port you want to view.

2 Press the <Space Bar> to select it.

3 Press [View/Modify].

The View Data Port screen (for the selected printer) is displayed.

4 Press [Exit] to return to the Data Port Configuration screen.

MMLink Data Port

The View Data Port screen for Meridian ACCESS Link (Figure 9-10) allows you to view link characteristics.

Figure 9-10xxx
View Data Port screen (MMLink)

```
                        Hardware Administration

View Data Port

Data Port Location:           1-3-2

Device Type:                  MMLink

Device Name:                  CONSOLE

Baud Rate:                    1200 [2400] 4800 9600

Parity:                       Even Odd [None]

Number of Windows:            1

Window Width:                 80

Window Height:                24

  Exit
```

The following read-only fields are displayed in the screen:

- ***Data Port Location*** - The location (node-card-port) in the system.
- ***Device Type*** - The function of the port. It will be set to “MMLink”.
- ***Device Name*** - The name of the device.
- ***Baud Rate*** - Set this field to “2400” for MMLink.
- ***Parity*** - This field is not used for MMLink.
- ***Number of Windows*** - This field specifies the number of windows that can be used simultaneously. This will be “1” for ACCESS.
- ***Window Width*** - This field specifies the window width used.
- ***Window Height*** - This field specifies the window height used.

Procedure 9-8xxx
Viewing the MMLink data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the MMLink data port you want to view.

2 Press the <Space Bar> to select it.

3 Press [View/Modify].

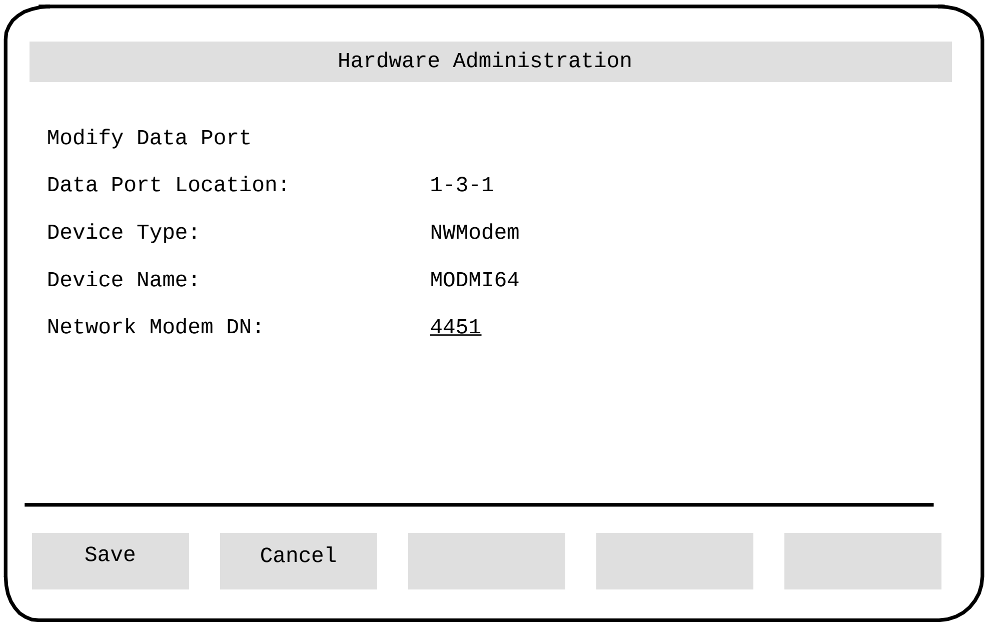
The View Data Port screen (for the selected MMLink) is displayed.

4 Press [Exit] to return to the Data Port Configuration screen.

NWModem Data Port

The Modify Data Port screen for Networking Modems (Figure 9-11) allows you to specify the Directory Number (DN) of the modem connected to the selected port.

Figure 9-11xxx
Modify Data Port screen (NWModem)



The following fields are displayed on this screen:

- ***Data Port Locatio***n - The physical location (node-card-port) in the system.
- ***Device Type*** - The function of the port. This will be “NWModem”.
- ***Network Modem DN*** - The directory number (up to 8 digits) used to identify the modem connected to the port. This field can be modified.

Procedure 9-9xxx
Modifying the NWModem data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the NWModem data port you want to modify.

2 Press the <Space Bar> to select it.

3 Press [View/Modify].

The View Data Port screen (for the selected NWModem) is displayed.

4 Use [Save] to save any changes or [Cancel] to disregard any changes.

The Data Port Configuration screen is displayed.

SMDI Data Port

The View Data Port screen for SMDI (Figure 9-12) allows you to view the baud rate, parity, and transmit mode of the serial connection to the SL-100 or DMS switch at the selected port.

Figure 9-12xxx
View Data Port screen (SMDI)

```
                    Hardware Administration
View Data Port

Data Port Location:           13-1-3

Device Type:                  SMDI

Device Name:                  CON183

Baud Rate:                    1200 [2400] 4800 9600

Parity:                       [Even] Odd  None

Transmit Mode:                Duplex

Link name:                    Link1 *

   Exit
```

*   The link name used here is for illustration purposes only.

The following read-only fields are displayed in the screen:

- ***Data Port Location*** - The physical location (node-card-port) in the system.
- ***Device Type*** - The function of the port. This will be “SMDI”.
- ***Device Name*** - The name of the device.
- ***Baud Rate*** - Set this field to “2400” for the MPC card or “1200” for the 1X67FA card.
- ***Parity*** - This will be “Even”.
- ***Transmit Mode*** - This will be “Duplex”.
- ***Link Name -*** The name of the link as defined during system installation. It is not recommended that this name be changed once users have been set up. See the section “Modify Hardware” in Appendix A, “System Administration Tools” for more information.

Procedure 9-10xxx
Viewing the SMDI data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the SMDI data port you want to view.

2 Press the <Space Bar> to select it.

3 Press [View/Modify].

The View Data Port screen (for the selected SMDI link) is displayed.

4 Press [Exit] to return to the Data Port Configuration screen.

Printing node or data port information

The following procedure describes how to print a list of all the node or data port information contained in the hardware database.

Figure 9-13xxx
The Hardware Administration menu

```
                    Hardware Administration

                  1  Node Configuration

                  2  T1 Link Configuration

                  3  Data Port Configuration

                  4  Print All Node Information

                  5  Print All Data Port Information

Select an item >

   Exit        *Cancel                     *Continue
               Printing                    Printing
```

* The Printing softkeys appear after item 4 or 5 has been selected.

Procedure 9-11xxx
Printing node and data port information

Starting point : The Hardware Administration menu, item <4> or <5> selected.

The following softkeys appear: [Continue Printing] and [Cancel Printing].

You are prompted to check that the printer is ready and on-line.

1 Choose step 1a to print the node or data port information or 1b to cancel.

a. Use [Continue Printing].

The node or data port information begins printing.

Once printing is complete, the Hardware Administration menu and its softkeys are redisplayed; you may stop printing at any time by proceeding to 2b.

b. Use [Cancel Printing].

The print operation is cancelled and you are returned to the Hardware Administration menu.

There may be some delay before control is returned to the screen while the system waits for the printer to stop printing.

System Status and Maintenance

The System Status and Maintenance function provides monitoring and control screens through which you obtain views of the operational state of the system at eight levels: system, nodes, cards, T1 links, SMDI links, T1 channels, DSP ports and disks. The System Status and Maintenance functions are used in the course of routine maintenance, and allow you to take any component of the system out of service while performing maintenance. A component can be taken out of service by disabling it (forcing it out of its operational state), or by performing a courtesy disable, which progressively disables active ports as they become idle. The Courtesy Disable avoids any disruption of calls in progress. The following maintenance-related actions can be taken:

- ***System Courtesy Down*** - is for broad maintenance activities, such as reconfiguring the serving switch, which necessitates a power shutdown on the DMS VoiceMail system.
- ***Courtesy Disable Ports or (forced) Disable Node*** - the choice of which is dependent on the nature of the work to be carried out, and the state of the node (information about which is obtained through the System Status and Maintenance displays).
- ***Card Disable*** - is used before performing diagnostics on a card, such as for a card showing “Faulty” status on one of its ports.
- ***Courtesy Disable or (forced) Disable of DSP Ports*** - is used before performing tests on a port.
- ***Courtesy Disable or (forced) Disable of T1 Channels*** - is used before performing maintenance on a T1 channel.
- ***Disk Synchronization -*** is used after a faulty disk has been replaced with a new disk and needs to be resynchronized with its partner.

The System Status and Maintenance function also provides a facility to print SEERs, an integral part of service and maintenance activities.

The System Status and Maintenance menu

The System Status and Maintenance menu (Figure 10-1) provides ten options.

Figure 10-1xxx
The System Status and Maintenance menu

```
                    System Status and Maintenance

          1  System Status

          2  Node Status

          3  Card Status

          4  DSP Port Status

          5  T1 Link Status

          6  T1 Channel Status

          7  SMDI Link Status

          8  SPM Cross Reference Table *

          9  Disk Maintenance

         10  System Event and Error Reports
 ______________________________________________________________________

 Select an item >
    Exit                                  Silence
                                           Alarm
```

* In a CPE environment, the SPM is known as the MSM (Message Services Module).

Procedure 10-1xxx
Using the System Status and Maintenance menu

Starting point : The Main Menu, entered.

The System Status and Maintenance menu appears (Figure 10-1).

1 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears.

See the following sections for details:

<1> “System Status”
<2> “Node Status”
<3> “Card Status”
<4> “DSP Port Status”
<5> “T1 Link Status”
<6> “T1 Channel Status”
<7> “SMDI Link Status”
<8> “SPM Cross Reference Table”
<9> “Disk Maintenance”
<10> “System Event and Error Reports”

2 Use [Exit] to return to the Main Menu.

Silencing Alarms

When the system sounds an alarm, you may silence it using the [Silence Alarm] softkey. When this softkey is pressed, the softkeys displayed in Figure 10-2 are displayed.

Figure 10-2xxx
Alarm softkeys

```
                    System Status and Maintenance


          1  System Status
          2  Node Status
          3  Card Status
          4  DSP Port Status
          5  T1 Link Status
          6  T1 Channel Status
          7  SMDI Link Status
          8  SPM Cross Reference Table *
          9  Disk Maintenance
          10System Event and Error Reports
______________________________________________________________________
Select an item >
              Cancel      Silence          Silence       Silence
                          Critical Alarm   Major Alarm   Minor Alarm
```

* In a CPE environment, the SPM is known as the MSM (Message Services Module).

An alarm will sound if the corresponding severity level SEER is issued indicating that a problem exists. By using the appropriate softkey you can silence either critical, major, or minor alarms. The [Cancel] softkey causes the original set of softkeys to be displayed without silencing any alarms. Try to clear the problem as well or the alarm could be turned on again if you simply silence it. Alarms persist until you silence them (there is no timeout period after which they are turned off by the system.)

System Status

The System Status screen (Figure 10-3) allows you to view the operational status of the system, manipulate the status of the nodes, and courtesy down the system or individual nodes. This screen is identical to the System Status screen displayed from the Logon screen, with the exception being that the Logon System Status screen does not have the Courtesy Down System softkey.

Figure 10-3xxx
System Status screen

```
                     System Status and Maintenance

System Status:   InService               Alarm Status:  Critical=Off Major=Off Minor=Off
Last Event: 60-00 PRM:  All System Programs Started   4/19  16:31

                                   DSP Port/Channel Status                Storage Used
Node Type    Node Status    Active Idle OutSv  Faulty PendingOthers       Voice    Text
  1  MSP     InService
  2  MSP     InService
  3  SPN     InService        0    24    0        0       0      0         43%      5%
  4  SPN     InService        24   0     0        0       0      0         30%      24%
  5  SPN     Takeover         0    24    0        0       0      0         30%      32%
  6  SPN     OutOfService     6    18    0        0       0      0         17%      40%
  7  SPN     Shutting Down    0    0     12       0       11     1         48%      32%
  8  SPN     Faulty           0    0     24       0       0      0         56%      35%
  9  SPN     Loading          0    15    8        0       1      0         12%      10%
 10  SPN     TakeOver         0    24    0        0       0      0         26%      18%
 13  TIFN    InService        0    72    24       0       0      0
 14  TIFN    OutOfService     0    0     0        0       0      0
 15  TIFN    InService        0    48    0        0       0      0
 16  TIFN    InService        0    48    0        0       0      0

Select a softkey >
     Exit                                                            Courtesy Down
                                                                        System*
```

*When the system is down, this softkey becomes [Activate System].

The following fields are displayed in the System Status screen:

- ***System Status*** - This field displays the current system status. Your system can be in one of the following states:
  - ***InService*** indicates that all critical programs on all nodes are operational and the system is accepting calls.
  - ***CourtesyPending*** indicates that the system is in the process of shutting down. This occurs after using the [Courtesy Down System] softkey. Incoming calls are directed to an attendant. Calls in progress are not interrupted. Each port is disabled as it becomes idle. The software remains loaded.

- ***CourtesyDown*** indicates that the system has shut down and is no longer operational nor accepting calls. The software remains loaded. When the system is down, the [Courtesy Down System] softkey becomes [Activate System]. When used, the system will restart and eventually return to an InService state.
- ***Loading*** indicates that the system is loading software while booting up.

- ***Alarm Status -*** This field indicates the state of each of the following alarm categories:

  Critical alarms indicate a service-affecting problem that requires immediate attention.

  Major alarms indicate a service-threatening problem that may be allowed to persist (for up to 24 hours). If not attended to, the alarm will become critical.

  Minor alarms indicate a problem that has no impact on the system or users.

  The status for each type of alarm will be one of the following:

  - ***Off*** indicates that there are no new alarms. This does not necessarily mean that there are no error conditions as alarms may have been silenced from the Logon screen, but the error conditions causing the alarm may still exist.
  - ***On*** indicates that one or more alarm situations was detected.
  - ***Unk*** indicates that the status is unknown.

- ***Last Event -*** The most recent system event or error (SEER) logged.
- ***Node -*** The node to which the following measurements apply.
- ***Type -*** The type of node.
- ***Node Status -*** The status of the nodes in your system. The node types include the MSP (Multi-Server Processor), SPN (Signal Processing Node) and TIFN nodes (Telephony Interface Node). A node may be in one of the following states:

  - ***InService*** indicates that node is running and accepting calls. For the MSP node, it indicates that node is running.
  - ***Unequipped*** indicates that the node is not defined in the hardware database. *The System Administration Tools* chapter in the Appendix describes how to modify the hardware database.
  - ***Faulty*** indicates that a hardware problem is detected.
  - ***Loading*** indicates that the node is currently starting up and loading software into memory. No software is running when the node is in this state.

- *InSvStandby* (used with the TIFN or MSP node) indicates that the node is running and is ready to take over operations for the paired redundant node.
- *OutOfService* indicates that the node is no longer operational, as a result of a courtesy disable or forced disable.
- *Booting* indicates that an operating system is being loaded on to the node.

- ***DSP Port Status*** - These fields reflect the state of each DSP port on the associated SPN node. For each port that is in a particular state, an entry is made in the appropriate column. A DSP port may be in one of the following states:
  - ***Active*** indicates that the port is operational and is currently in use.
  - ***Idle*** indicates that the port is operational but not in use at the moment. The port is ready to accept calls.
  - ***OutSv*** indicates that the associated port is not operational, and is not accepting calls.
  - ***Faulty*** indicates that the a hardware problem has been detected in the DSP port.
  - ***Pending*** indicates that there has been a request to shut down the port. The port is still active, pending an active call being disconnected before shutting down.
  - ***Other*** indicates that the port is temporarily unavailable. This usually occurs while the system is booting up. The status remains as “Other” while the software is loading. Once fully loaded, the status automatically becomes “Idle”. The status may also appear as “Other” when you re-enable a port (for as long as the necessary software is loading). The status returns to “Idle” once the port has been enabled.
- ***Storage Used*** - This field indicates the amount of voice and text storage used as a percentage of available storage on the user volume of this node. (If the disk on a node is bad, percentages are not displayed.) It is only valid for the SPN node.

Procedure 10-2xxx
Courtesying down the system

Starting point : The System Status and Maintenance menu, <1> entered.

The System Status screen appears (Figure 10-3).

1 Choose step 1a to courtesy down the system, or 1b to return to the System Status and Maintenance menu.

a. Use [Courtesy Down System].

[Activate System] replaces [Courtesy Down System].

The system may take some time in disabling the system since it waits for all active DSP ports on all nodes to become idle; the message "WORKING ..." will be displayed during this interval.

If a DSP port does not become idle during a courtesy down, disable the DSP Port manually by following the procedure described under "DSP Port Status". Wait a few minutes to ensure that an in-progress call is not dropped.

The system can be re-enabled at any time during the process by using [Activate System].

If a DSP port does not re-enable, enable it by following the procedures described under "DSP Port Status".

b. Use [Exit] to return to the System Status and Maintenance menu.

Node Status

The Node Status screen (Figure 10-4) displays the operational status of the nodes in your system. The softkeys displayed on this screen are used to enable and disable nodes on the system and courtesy disable ports on an SPN node in the system.

Figure 10-4xxx
Node Status screen

```
                    System Status and Maintenance

Node Status

System Status: InService  Alarm Status: Critical=Off Major=Off Minor=Off

Type    Node    Status                        Paired Node Status
 MSP      1     Inservice                            2    InSvStandby
 SPN      3     InService                            4    InService
 TIFN    13     InService                           14    InSvStandby

 *Select a softkey >

   Exit       Enable      Disable     Courtesy Dis-
              Node        Node        able Ports
```

```
* "Enter the number of the node you want to enable" appears when [Enable
  Node] is used.
* "Enter the number of the node you want to disable" appears when [Disable
  Node] is used.
* "Enter the number of the node you want to Courtesy Disable Ports" appears
  when [Courtesy Disable Ports] is used.
```

The following fields are displayed on the Node Status screen:

- ***System Status*** - This field displays the current system status. See the section "System Status" for details.
- ***Alarm Status*** - This field indicates whether there are any critical, major or minor alarms. See the section "System Status" for details.
- ***Type*** - The type of node. The three types of nodes are:
  - ***MSP*** or Multi-Server Processor
  - ***SPN*** or Signal Processing Node
  - ***TIFN*** or Telephony Interface Node
- ***Node*** - The node number to which the following measurements apply.
- ***Status*** - The node will be in one of the following states:

- ***InService*** indicates that the node is running and accepting calls. For the MSP node, it indicates that node is running and is in load-sharing mode with its paired redundant MSP node.
- ***Unequipped*** indicates that the node is not defined in the hardware database. *The System Administration Tools* chapter in the Appendix describes how to modify the hardware database.
- ***Faulty*** indicates that a hardware problem is detected.
- ***Loading*** indicates that the node is currently starting up and loading software into memory. No software is running when the node is in this state.
- ***InSvStandby*** (used with the TIFN and MSP node) indicates that the node is running and is take over for the paired redundant node.
- ***InSvAlarm*** indicates that at least one of the links on the (TIFN) node is faulty.
- ***OutOfService*** indicates that the node is no longer operational, as a result of a forced disable.
- ***Booting*** indicates that an operating system is being loaded on to the node.

- ***Paired Node* -** The number of the node which is paired with the original node.
- ***Status* -** The status of the paired node. See the above descriptions.

Procedure 10-3xxx
Enabling and disabling nodes

Starting point : The System Status and Maintenance menu, <2> entered.

The Node Status screen appears (Figure 10-4).

1 Choose step 1a to enable a node, 1b to disable a node, 1c to courtesy disable ports on a node, 1d to return to the System Status and Maintenance menu.

a. Use [Enable Node].

You are prompted for the node number.

Enter the required number followed by <Return>.

Note 1: *If you have just disabled a node and are re-enabling it, wait 3 to 5 minutes after using [Disable Node] before you use [Enable Node].*

Note 2: *If you enable MSP1 when it is in InSvStandby, an MSP switchover will occur.*

The system may take some time in enabling the node.

b. Before you disable a node, it is suggested you use [Courtesy Disable Ports] on the affected node. Once the ports are disabled, press the [Disable Node] softkey.

   You are prompted for the number of an in-service node.

   Enter the node number followed by <Return>.

   The system may take some time in disabling the node.

   If you are going to re-enable the node, wait 3 to 5 minutes before using [Enable Node].

c. Use [Courtesy Disable Ports].

   You are prompted for the number of an in-service node.

   Enter the SPN node number followed by <Return>.

   The system may take some time in disabling all the ports on a node since it waits for all active DSP ports to become idle. The node itself will remain InService.

d. Use [Exit] to return to the System Status and Maintenance menu.

Card Status

The Card Status screen (Figure 10-5) displays the operational status of the cards in your system. The softkeys displayed on this screen are used to enable and disable VP and T1 cards on the SPN and TIFN nodes and initiate diagnostics when necessary. (To disable other types of cards, use [Disable Node] on the Node Status screen.)

Figure 10-5xxx
The Card Status screen

```
                      System Status and Maintenance

Card Status for Node 13

System Status: InService   Alarm Status:Critical=Off Major=Off Minor=Off

Card#     Location     Description                          Status
 1        13-1-*       T1                                   InService
 2        13-2-*       Empty                                Unequipped
 3        13-3-*       Enhanced Single Board Computer       InService

Select a softkey >

  Exit       Enable      Disable                   OutOfService
             Card        Card                      Diagnostics
```

The following fields are displayed on the Card Status screen:

- ***System Status*** - See the description in the section “System Status”.
- ***Alarm Status*** - See the description in the section “System Status”.
- ***Card Number*** - The number of each card on the selected node.
- ***Card Location*** - The physical location (node-card) of each card on the selected node.
- ***Card Description*** - Function of each card.
- ***Card Status*** - The current state of each card on the selected node.
  - ***Unequipped*** indicates that either the card slot is empty but defined in hardware database or the card is in the slot but not defined in the hardware database.
  - ***Faulty*** indicates that a hardware problem has been detected for the card.
  - ***InService*** indicates that the card is operational.

- ***OutOfService*** indicates that the card has been disabled.

Procedure 10-4xxx
Enabling and disabling cards

Starting point : The System Status and Maintenance menu, <3> entered.

System responds: Enter the node number for card status > __
Enter specified node.
The Card Status screen appears (Figure 10-5).

1 Choose step 1a to enable a card, 1b to disable a card, 1c to activate diagnostics on an out-of-service or faulty card, or 1d to return.

a. Use [Enable Card].

Note: *Only voice processor and T1 cards can be enabled from this screen. To enable other cards use [Enable Node] in the Node Status screen. See the previous section, "Node Status".*

You are prompted for the number of an out-of-service card.

Enter the card number followed by <Return>.

The system may take some time in enabling the card. The message "WORKING ..." will be displayed during this interval.

b. Use [Disable Card].

Note: *Only voice processor cards and T1 cards can be disabled from this screen. To disable other cards use [Disable Node] in the Node Status screen.*

You are prompted for the number of the card you want to disable.

Enter the card number followed by <Return>.

The system may take some time in disabling the card. The message "WORKING ..." will be displayed during this interval.

c. Use [OutOfService Diagnostics].

You are prompted for the number of an out-of-service or faulty card.

Enter the card number followed by <Return>.

The system may take some time in running diagnostics. The message "WORKING ..." will be displayed during this interval.

Note: *If the OutOfService Diagnostics fails the card will become faulty. If the card state is faulty you can run OutOfService diagnostics. If passed, the card is put into the OutOfService state.*

d. Use [Exit] to return to the System Status and Maintenance menu.

DSP Port Status

The DSP Port Status screen (Figure 10-6) allows you to view the operational status of the DSP ports on a node, manipulate their status, and courtesy disable individual ports when necessary.

Figure 10-6xxx
The DSP Port Status screen

```
                     System Status and Maintenance

DSP Port Status for Node 5

System Status: InService Alarm Status:Critical=Off Major=Off Minor=Off

Location          DSP Port Status

5-1-*              1-Idle    2-Active    3-Active    4-Idle
                   5-Idle    6-Active    7-Active    8-Idle
                   9-Idle   10-Active   11-Active   12-Idle

5-2-*             13-Idle   14-Active   15-Active   16-Idle
                  17-Idle   18-Active   19-Active   20-Idle
                  21-Idle   22-Active   23-Active   24-Idle

Select a softkey >

                                           Courtesy       Change to
  Exit     Enable Port   Disable Port   Disable Port     Range Mode
```

The following fields are displayed on the DSP Port Status screen:

- ***System Status*** - See the description in the section “System Status”.
- ***Alarm Status*** - See the description in the section “System Status”.
- ***Location*** - The physical location (node-card) of each port on the selected node. For example, “23-Active” indicates that the 23rd DSP port on the second card of the fifth node is active.
- ***DSP Port Status*** - The current state of each DSP port.
  - ***Active*** indicates that the DSP port is operational and in use.
  - ***Idle*** indicates that the DSP port is operational but not in use.
  - ***Faulty*** indicates that the system has detected an error.
  - ***OutOfService*** indicates that the port is no longer operational, as a result of a courtesy disable or forced disable.
  - ***POutService*** indicates that the port is in the process of shutting down, pending disconnection of an active call. The port is still active.

- ***UnEquipped*** indicates that the DSP port is not defined in the hardware database. For more information about modifying the hardware database, see *System Administration Tools* in Appendix A.
- ***No Resource*** indicates a transition state that occurs during the initial stages of software loading (after a request to enable a port). When software begins to load, the port is initially in this state, followed by Loading and finally, once the software has finished loading, Idle.
- ***Loading*** indicates that the DSP port is currently starting up after a request to enable and that the necessary software is loading.

If you need to enable, disable or courtesy disable a number of DSP ports, use the [Change to Range Mode] softkey first. (This only works with a contiguous range of ports. For example, it will work if you need to disable ports 3 to 7, but not if you need to disable ports 1, 3 and 7. When you to toggle to range mode, this softkey changes to [Change to Single Mode]. If you are in single mode, follow Procedure 10-5. If you are in range mode, follow Procedure 10-6.

Procedure 10-5xxx
Enabling and disabling DSP ports in single mode

Starting point : The System Status and Maintenance screen, <4> entered.

You are prompted for a node number.

1 Enter the number of the node on which the port resides.

The DSP Port Status screen appears (see Figure 10-6).

2 Choose step 2a to enable a DSP port, 2b to disable a DSP port, 2c to courtesy disable a DSP port, or 2d to exit the DSP Port Status screen.

a. Use [Enable Port].

You are prompted for the number of an out-of-service port.

Enter the port number followed by <Return>.

The system may take some time in enabling the DSP port. The message "WORKING ..." will be displayed during this interval.

b. Use [Disable Port].

You are prompted for the number of an in-service port.

Enter the port number followed by <Return>.

The system may take some time in disabling the DSP port. The message "WORKING ..." will be displayed during this interval.

c. Use [Courtesy Disable Port].

You are prompted for the number of an in-service or active DSP port.

Enter the port number followed by <Return>.

The system may take some time in disabling the port since it waits for the port to become idle before disabling it. The message "WORKING ..." will be displayed during this interval.

d. Use [Exit] to return to the System Status and Maintenance menu.

Procedure 10-6xxx
Enabling and disabling DSP ports in range mode

Starting point The System Status and Maintenance screen, <3> entered.

You are prompted for the node number.

1 Enter the number of the node on which the DSP port resides.

The DSP Port Status screen is displayed.

2 Choose step 2a to enable a range of DSP ports, 2b to disable a range of DSP ports, 2c to courtesy disable a range of DSP ports, or 2d to exit the DSP Port Status screen.

a. Use [Enable Port].

You are prompted for the number of the first DSP port in the range of ports you want to enable.

Enter the number of the first DSP port in the range followed by <Return>.

You are prompted for the number of the last DSP port in the range.

Enter the number of the last DSP port in the range followed by <Return>.

The system may take some time in enabling the DSP ports. The message "WORKING ..." will be displayed during this interval.

The system displays a message to inform you of the number of ports successfully enabled, and the number of ports that could not be enabled.

b. Use [Disable Port].

You are prompted for the number of the first DSP port in the range of ports you want to disable.

Enter the number of the first DSP port in the range followed by <Return>.

You are prompted for the number of the last DSP port in the range.

Enter the number of the last DSP port in the range followed by <Return>.

The system displays a message to inform you of the number of ports successfully disabled, and the number of ports that could not be disabled.

c. Use [Courtesy Disable Port].

You are prompted for the number of the first DSP port in the range of ports you want to courtesy disable.

Enter the number of the first DSP port in the range followed by <Return>.

You are prompted for the number of the last DSP port in the range.

Enter the number of the last DSP port in the range followed by <Return>.

The system may take some time in disabling the DSP port since it waits for the port to become idle before disabling it. The DSP port status will be POutOfService during this interval.

The system displays a message to inform you of the number of ports successfully courtesy disabled, and the number of ports that could not be courtesy disabled.

d. Use [Exit] to return to the System Status and Maintenance menu.

T1 Link Status

The T1 Link Status screen (Figure 10-7) allows you to view the operational status of the T1 links on the system, enable or disable a link, activate one of the clock reference candidates, and switch the link to its alternate connection when required.

Figure 10-7xxx
The T1 Link Status screen

```
                              System Status and Maintenance

T1 Link Status

System Status: InService    Alarm Status:Critical=Off Major=On Minor=Off

          Clocking    Primary Connection                 Redundant Connection
Link Cand Mode    T1    T1                          T1     T1
ID             Number  Location    Status           NumberLocation  Status
 A    Y          1     11-1-1      InService        5      12-1-1    InService
 B               2     11-1-2      InService        6      12-1-2    OutOfService
 C               3     11-1-3      InService        7      12-1-3    Standby
 D               4     11-1-4      OutOfService     8      12-1-4    InService
 E         R     9     13-1-1      InService        13     14-1-1    InService
 F              10     13-1-2      Faulty           14     14-1-2    InService
 G              11     13-1-3      InService        15     14-1-3    Standby
 H    Y         12     13-1-4      InService        16     14-1-4    Faulty

 Select a softkey >

   Exit          Enable T1      Disable T1      Change T1         Switch
                                              Clocking Mode        Link
```

The following fields are displayed on the T1 Link Status screen:

- ***System Status*** - See the description in the section “System Status”.
- ***Alarm Status*** - See the description in the section “System Status”.
- ***Link ID*** - An alphabetic designation used to identify the T1 link in your system. This corresponds to the Link ID in the T1 Link Configuration screen in Hardware Administration.
- ***Cand*** - This is a read-only field. A “Y” in this field indicates that the link has been nominated as a candidate for clock referencing. A candidate is nominated from the T1 Link Configuration screen in Hardware Administration. See the section “Modifying the T1 link setup” in the “Hardware Administration” chapter for more information.
- ***Clocking Mode*** - The currently activated clock reference is indicated with an “R” in this field. A link is activated by using the [Change T1 Clocking Mode] softkey as described in Procedure 10-7. If none of the

links are activated as the clock reference, the system is in free-run mode, meaning that the system is using the internal SPM (MSM) clock.

- ***Primary Connection T1 Number -*** The number of the primary T1 connection within the specified T1 link.
- ***Primary Connection T1 Location -*** The location of the primary T1 connection in the system. This number represents the location in terms of the node-card-span.
- ***Primary Connection Status -*** The current state of the primary T1 connection.
  - ***Unequipped*** indicates that the connection is not defined in the hardware database. For more information about modifying the hardware database, see *System Administration Tools* in the Appendix.
  - ***Faulty*** indicates that a hardware problem has been detected on the connection.
  - ***Alarm*** indicates that an alarm has been detected on the link.
  - ***InService*** indicates that the T1 connection is fully operational and is currently accepting calls.
  - ***InSvStandby*** indicates that the connection is not currently taking calls but is ready to accept calls for the paired T1 connection on the same T1 link.
  - ***OutOfService*** indicates that the connection is not operational due to a forced disable.
  - ***Pending*** indicates that the connection is in the process of shutting down or restarting.
- ***Secondary Connection T1 Number -*** The number of the secondary T1 connection within the specified T1 link.
- ***Secondary Connection T1 Location -*** The location of the secondary T1 connection in the system. This number represents the location in terms of the node-card-span.
- ***Secondary Connection Status -*** The current state of the secondary connection. See the descriptions for the *Primary Connection Status*.

You may perform the following actions on T1 connections:

- ***Disable T1 -*** When a T1 connection is disabled it is no longer used to accept calls. (This action is not allowed when the connection status is "Unequipped".) Once the connection is disabled, its status becomes OutOfService.
- ***Enable T1 -*** This action starts up a T1 connection that is currently in an OutOfService state. Once the connection is fully enabled, its status becomes InService if the paired T1 connection is not InService, or InSvStandby if the paired T1 connection is already InService.

- ***Change T1 Clocking Mode*** - This action allows you to activate one of the nominated links as the clock reference. Alternatively, you can place the system in free-run mode (in which case the internal SPM (MSM) clock is used instead of an external reference provider).
- ***Switch Link*** **-** This action allows you to switch from an InService T1 connection to the paired InSvStandby T1 connection. This switching is allowed only if one T1 connection (of a pair) is InService and its partner is InSvStandby.

  Note: Only one of the paired T1 connections can be InService at any one time.

Procedure 10-7xxx
Enabling, disabling and switching T1 connections

Starting Point : The System Status and Maintenance menu, <5> entered.

The T1 Link Status screen is displayed (Figure 10-7).

1 Choose step 1a to enable a T1 connection, 1b to disable a T1 connection, 1c to change the T1 clock reference, 1d to switch the T1 link, or 1e to exit the T1 Link Status screen.

a. Use [Enable T1].

You are prompted for the T1 number.

Enter the T1 number followed by <Return>.

The system may take some time in enabling the T1 connection.

b. Use [Disable T1].

You are prompted for the T1 number of a connection.

Enter the T1 number followed by <Return>.

The system may take some time in disabling the T1 connection.

c. Use [Change T1 Clocking Mode].

You are prompted for the Link ID.

Enter the Link ID followed by <Return>. Alternatively, you can enter **Z** followed by <Return> (for free run mode).

The specified link ID becomes the new clock reference. If another link was previously activated, it is deactivated as only one link can serve as the reference. If you entered Z, a previously activated link is deactivated.

d. Use [Switch Link].

You are prompted for the Link ID.

Enter the Link ID followed by <Return>.

This changes the status of the primary and redundant connections from InSvstandby to in-service and vice versa.

e. Use [Exit] to return to the System Status and Maintenance menu.

T1 Channel Status

The T1 Channel Status screen (Figure 10-8) allows you to view the operational status of the T1 channels in the system, manipulate their status, and courtesy disable individual channels when necessary.

Figure 10-8xxx
The T1 Channel Status screen

```
                     System Status and Maintenance
T1 Channel Status

System Status:   InService      Alarm Status:  Critical=Off Major=Off Minor=Off

       Channels
Link   1 2  3 4 5 6 7 8  9  10 11  12 13  14 15  16 17  18 19  20 21 22 23 24
 A     a a .  a . . . .  .  . . .  a  .   a  a   .  .   .  .   .  .  .  .  .
 B     a a .  a o o o o  .  . . .  .  .   a  a   .  .   .  a   .  .  .  .  .
 C     . a .  a . . . .  .  . . .  a  .   a  .   .  .   .  .   .  .  .  .  .
 D     a a .  a . . . .  .  . . .  a  .   a  a   .  .   .  .   .  .  a  .  .
 E     a . .    o o o o  a  . . .  .  .   .  .   .  .   .  .   .  .  .  .  .
 F     . a .  a . . . a  .  . . .  a  .   a  a   .  .   .  .   .  .  .  .  .
 G     a a .  . . . . .  .  . . .  a  .   .  .   .  .   .  .   .  .  .  .  .
 H     a a .  a . . . .  .  . . .  a  .   a  a   .  F   F  F   F  .  .  .  .

a = Active/In Use        . = Idle        O  = Out of Service      R  = No Resources
F = Faulty               P  = Pending   space= Unequipped          L = Loading
C = CourtesyDown

Select a softkey >

Exit | Enable Channel | Disable Channel | Courtesy Disable Channel | Change to Range Mode
```

The following fields are displayed on the T1 Channel Status screen:

- ***System Status*** - See the description in the section “System Status”.
- ***Alarm Status -*** See the description in the section “System Status”.
- ***Link*** - The ID of the T1 link. This is an alphabetic character.
- ***Channel Status*** - The current state of each channel.
  - ***Active/In Use*** indicates that the T1 channel is operational and in use.
  - ***Idle*** indicates that the channel is operational but not currently in use.
  - ***OutOfService*** indicates that the channel is no longer operational.
  - ***No Resources*** indicates that the T1 channel is available, but there is no software associated with it.
  - ***Faulty*** indicates that the system has detected an error in the channel.
  - ***Pending*** indicates that the channel is in the process of shutting down or restarting.

- ***UnEquipped*** indicates that the channel is not defined in the hardware database. For more information about modifying the hardware database, see *System Administration Tools* in the Appendix.
- ***Courtesy Down*** indicates that the channel is in a courtesy down state as a result of performing a Courtesy Down System. The channel does not accept calls in this state.

If you need to enable, disable or courtesy disable a number of T1 channels, use the [Change to Range Mode] softkey first. (This only works with a contiguous range of channels. For example, it will work if you need to disable T1 channels 3 to 7, but not if you need to disable T1 channels 1, 3 and 7. When you to toggle to range mode, this softkey changes to [Change to Single Mode].

If you are in single mode, follow Procedure 10-8. If you are in range mode, follow Procedure 10-9.

Procedure 10-8xxx
Enabling and disabling T1 channels in single mode

Starting point : The System Status and Maintenance screen, <6> entered.

The T1 Channel Status screen appears (see Figure 10-8).

1 Choose step 2a to enable a channel, 2b to disable a channel, 2c to courtesy disable a channel, or 2d to exit the T1 Channel Status screen.

a. Use [Enable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of an out-of-service channel.

Enter the channel number followed by <Return>.

The system may take some time in enabling the channel. The message "WORKING ..." will be displayed during this interval.

b. Use [Disable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of an in-service channel.

Enter the channel number followed by <Return>.

The system may take some time in disabling the channel. The message "WORKING ..." will be displayed during this interval.

c. Use [Courtesy Disable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of an in-service channel.

Enter the channel number followed by <Return>.

The system may take some time in disabling the channel since it waits for the channel to become idle before disabling it.
The message “WORKING ...” will be displayed during this interval.

d. Use [Exit] to return to the System Status and Maintenance menu.

Procedure 10-9xxx
Enabling, disabling and switching T1 channels in range mode

Starting Point : The System Status and Maintenance menu, <6> entered.

The T1 Channel Status screen is displayed (see Figure 10-8).

1 Choose step 1a to enable a range of T1 channels, 1b to disable a range of T1 channels, 1c to switch a range of T1 channels, or 1d to exit the T1 Channel Status screen.

a. Use [Enable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of the first channel in the range of channels that you want to enable.

Enter the first channel number followed by <Return>.

You are prompted for the number of the last channel in the range of channels that you want to enable.

Enter the last channel number followed by <Return>.

The system may take some time in enabling the channels. The message “WORKING ...” will be displayed during this interval.

b. Use [Disable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of the first channel in the range of channels that you want to disable.

Enter the first channel number followed by <Return>.

You are prompted for the number of the last channel in the range of channels that you want to disable.

Enter the last channel number followed by <Return>.

The system may take some time in disabling the channels. The message “WORKING ...” will be displayed during this interval.

c. Use [Courtesy Disable Channel].

You are prompted for the link ID.

Enter the letter designation of the link followed by <Return>.

You are prompted for the number of the first channel in the range of channels that you want to courtesy disable.

Enter the first channel number followed by <Return>.

You are prompted for the number of the last channel in the range of channels that you want to courtesy disable.

Enter the last channel number followed by <Return>.

The system may take some time in disabling the channels since it waits for the channels to become idle before disabling them.
The message "WORKING ..." will be displayed during this interval.

d. Use [Exit] to return to the System Status and Maintenance menu.

SMDI Link Status

This screen displays the SMDI links in the system and the status of the primary and secondary ports on those links.

Figure 10-9xxx
The SMDI Link Status screen

```
                    System Status and Maintenance

SMDI Link Status

System Status: InService   Alarm Status:Critical=Off Major=Off Minor=Off

              Primary Link                         Redundant Link
Link     SMDI    SMDI                         SMDI    SMDI
ID      NumberLocation Status                Number   Location* Status
 A        1    11-1-1  InService               5      12-1-1    Standby
 B        2    11-1-2  InService               6      12-1-2    OutOfService
 C        3    11-1-3  InService               7      12-1-3    Standby
 D        4    11-1-4  OutOfService            8      12-1-4    InService
 E        9    13-1-1  Standby                13      14-1-1    InService
 F       10    13-1-2  Faulty                 14      14-1-2    InService
 G       11    13-1-3  InService              15      14-1-3    Standby
 H       12    13-1-4  InService              16      14-1-4    Faulty

 Select a softkey >
   Exit      Enable SMDI   Disable SMDI                 Switch
                                                         Link
```

* The Location field is blank if your system does not have redundant ports.

The following fields are displayed on the SMDI Link Status screen:

- ***System Status*** - See the description in the section “System Status”.
- ***Alarm Status*** - See the description in the section “System Status”.
- ***Link ID*** - An alphabetic designation used to identify the SMDI link in your system.
- ***SMDI Number*** - The specific SMDI port at which the SMDI link from the serving switch terminates on the SPM. There are two SMDI ports (one redundant) associated with each SMDI link.
- ***SMDI Location*** - The location (node-card-port) of the port in the system.
- ***Status*** - The current state of the SMDI port.
  - ***Unequipped*** indicates that the port is not defined in the hardware database. For more information about modifying the hardware database, see *System Administration Tools* in Appendix A.
  - ***Faulty*** indicates that a hardware problem has been detected on the port.

- ***InSvY elAlarm*** indicates that the SMDI port is in service but has lost the modem connection.
- ***InSvRedAlarm*** indicates that the SMDI port has lost the signaling with the DMS host.
- ***InService*** indicates that the SMDI port is fully operational and is currently accepting calls.
- ***InSvStandby*** indicates that the port is not currently taking calls but is ready to accept calls for the paired SMDI port on the same SMDI link.
- ***OutOfService*** indicates that the port is not operational, due to a forced disable, and is not accepting calls.
- ***Pending*** indicates that the port is in the process of shutting down or restarting.
- ***No Resource*** indicates a transition state that occurs during the initial stages of software loading (after a request to enable an SMDI link). When software begins to load, the link is initially in this state, followed by Loading and finally, once the software has finished loading, Idle.

You may perform the following actions on SMDI ports:

- ***Disable SMDI*** **-** When an SMDI port is disabled, the in call detail information no longer accompanies the call and the SMDI port status becomes OutOfService.

  Calls that are already in the UCD queue when the SMDI port is disabled will get the default service (i.e., the service associated with the primary UCD queue, namely voice messaging). Any new calls will also get the the default service.

  If this is not acceptable, disable the associated telephony channels before you disable the SMDI port. This will log out the UCD agents and, depending on how the UCD group is datafilled, calls can be routed for alternative treatment. For example, if the system has multiple SMDI links, calls could be routed to another UCD group. If you use this method, enable the associated channels before reenabling the SMDI port.

- ***Enable SMDI*** **-** This action starts up an SMDI port that is currently OutOfService. Once the port is fully enabled, its status becomes InService if the paired SMDI port is not InService, or InSvStandby if the paired SMDI port is already InService.
- ***Switch Link*** **-** This action is only possible if your system has redundant ports. It allows you to switch from an InService SMDI port to the paired InSvStandby SMDI port. This switching is allowed only if one SMDI port is InService and its partner is InSvStandby.

  Note: Only one of the SMDI ports within a pair can be InService at any one time. (Multiple pairs can be InService at the same time.)

Procedure 10-10xxx
Enabling, disabling and switching SMDI Links

Starting Point : The System Status and Maintenance menu, <4> entered.

The SMDI Link Status screen is displayed (Figure 10-9).

1 Choose step 1a to enable an SMDI port, 1b to disable an SMDI port, 1c to switch the SMDI link, or 1d to exit the SMDI Link Status screen.

a. Use [Enable SMDI].

You are prompted for the number of the SMDI port.

Enter the SMDI number followed by <Return>.

The system may take some time in enabling the port.

b. Use [Disable SMDI].

(If necessary, disable the associated channels first. See the description of Disable SMDI on the previous page for details.)

You are prompted for the number of an in-service SMDI port.

Enter the SMDI number followed by <Return>.

The system may take some time in disabling the port.

c. Use [Switch Link].

You are prompted for the Link ID.

Enter the Link ID followed by <Return>.

This changes the status of the primary and redundant ports from standby to in-service and vice versa.

d. Use [Exit] to return to the System Status and Maintenance menu.

SPM Cross Reference Table

The SPM Cross Reference Table (Figure 10-10) allows you to look up the card number (part number), shelf and slot for each card in the system.

Note: The SPM is known as the MSM on CPE systems.

Figure 10-10xxx
The SPM Cross Reference Table

```
System Status and Maintenance

SPM Cross Reference Table

Node Card   Card Type   CardNumber    Shelf    Slot
 1    1      Filler      NT9X19AA      26       07F
 1    2      SBC         NTGX05AA      26       08F
 1    3      Bus         NTGX10AA      26       09F
 2    1      Bus         NTGX10AA      26       10F
 2    2      Filler      NT9X19AA      26       11F
 2    3      SBC         NTGX05AA      26       12F
 3    1      SBC         NTGX05AA      26       30F
 3    2      VP12        NTGX12AA      26       31F
 3    3      VP12        NTGX12AA      26       32F
 4    1      VP12        NTGX12AA      26       27F
 4    2      VP12        NTGX12AA      26       28F
 4    3      SBC         NTGX05AA      26       29F
 5    1      VP12        NTGX12AA      26       17F
 5    2      VP12        NTGX12AA      26       18F
                                                      MORE BELOW
 Select a softkey >

   Exit
```

The following fields are displayed:

- ***Node -*** The node on which the card resides.
- ***Card -*** The card number.
- ***Card Type -*** Examples of card types as shown in Figure 10-10 are:
  - ***SBC*** - a single board computer (also known as the 68K card).
  - ***Bus*** - high-speed bus
  - ***VP12 -*** 12-channel voice processor
  - ***Filler -*** an empty card slot

  Other examples include the T1 transition module, 68k transition module, modem transition module, and bus controller transition module.
- ***CardNumber -*** The part number of the card.
- ***Shelf -*** The shelf on which the card is located.
- ***Slot -*** The slot in which the card resides. “F” indicates front. “R” indicates rear.

Disk maintenance

Disks are added to DMS VoiceMail in pairs. When new data is written to disk, both drives in a pair are updated at the same time with the same information. If one of the drives in a pair fails, it can be removed from service and replaced without loss of data or interruption of service.

When a disk fails due to any sort of SCSI error, the system automatically takes it out of service (puts it in “No Access” state) and generates a SEER. The shadowed disk continues to function and there is no service interruption. However, the failed disk should be replaced as soon as possible. You may also have to replace (or repair) a disk that has reported a large number of recovered errors. In the second case, you will have to take the disk out of service manually before replacing it. After a disk has been replaced or repaired, you will have to perform a disk synchronization in order to bring the paired disks in line with each other.

To replace a failed disk:

1 Disable the drive (if not already in No Access state). This disables the drive and puts it in No Access so that it can be replaced or repaired. See Procedure 10-11.
2 Replace the failed drive.
3 Resynchronize the disks. See Procedure 10-12.

The disk maintenance screen

The Disk Maintenance screen (Figure 10-11) shows the status of each disk pair in the system. The three possible states for a disk pair are “InSynch”, “Synching” and “OutofSynch”. If a SEER has alerted you to the fact that the system has automatically taken a disk out of service, check the Disk Maintenance screen to determine which pair is out of synch.

Figure 10-11xxx
The Disk Maintenance screen

```
                       System Status and Maintenance

Disk Maintenance

System Status: InService   Alarm Status: Critical=Off  Major=On  Minor=Off

Pair Number Node Type    Status

   1         MSP         InSynch
   2         SPN         Synching     34% done
   3         SPN         OutofSynch
   4         SPN         Insynch
   5         SPN         InSynch
   6         SPN         InSynch
   7         SPN         InSynch
   8         SPN         InSynch
   9         SPN         InSynch

Select a softkey >
   Exit      View Disk
               Pair
```

The following fields are displayed on this screen:

- ***System Status -*** This field displays the current system status. See the section “System Status” for a description of possible system states.
- ***Alarm Status -*** This field indicates whether or not there are any Critical, Major or Minor alarm. See the section “System Status” for a description of possible alarm states.
- ***Pair Number -*** The number of each disk pair in the system.
- ***Node Type -*** The type of node on which the pair resides.
- ***Status -*** The synchronization status. A disk pair can be in one of the following states:
  - ***InSynch*** indicates that both disks are operational and in synch with each other.
  - ***Synching*** indicates that the disks are currently synching (i.e., after pressing [Enable] in the Disk Pair Status screen).

- ***OutofSynch*** indicates that one of the disks is NoAccess and consequently out of synch with its shadowed pair. This happens if the system automatically puts a bad disk in No Access or if you disable the disk in order to replace or repair it.
- ***SynchIntrpted*** indicates that a disk synchronization operation has been interrupted. To recover from this state:
  1. Log on at the Tools level.
  2. Select Synchronize Disks.
  3. Run the init command on one of the nodes that the disk pair belongs to.
  4. Return to System Status and Maintenance, the Node Status screen.
  5. Disable and then reenable one of the nodes the disk pair belongs to.
  6. Return to Disk Maintenance and try synching the disk pair again.

When you press the [View Disk Pair] softkey, the screen shown in Figure 10-12 is displayed.

Figure 10-12xxx
The Disk Pair Status screen

```
                        System Status and Maintenance

Disk Pair Status for Pair X

System Status: InService   Alarm Status: Critical=Off  Major=On  Minor=Off

Node    Primary     Shadow

   3    ReadWrite   ReadWrite
   4    ReadWrite   ReadWrite

Select a softkey >

   Exit        Enable        Disable
```

The following fields are displayed on this screen:

- ***System Status -*** The current system status. See the section “System Status” for a description of possible system states.

- ***Alarm Status* -** Indicates whether there are any critical, major or minor alarms. See the section “System Status” for a description of possible alarm states.
- ***Node* -** The node on which the disk resides.
- ***Primary* -** This field indicates the status of the primary disk. A disk may be in one of the following states:
  - ***ReadWrite*** indicates that the disk is currently being read and written to. A disk that is in this state is operating normally.
  - ***NoAccess*** indicates that the disk is not being read or written to due to an error condition or a manual disable.
  - ***SynchSource,*** during a disk synch, indicates that the disk is the source of a disk synchronization.
  - ***SynchDest***, during a disk synch, indicates that the disk is the destination of a disk synchronization.
- ***Shadow* -** This field indicates the status of the shadowed disk. A disk may be in one of the following states:
  - ***ReadWrite*** indicates that the disk is currently being read and written to.
  - ***NoAccess*** indicates that the disk is not being read or written to due to an error condition or a manual disable.
  - ***SynchSource***, during a disk synch, indicates that the disk is the source of a disk synchronization.
  - ***SyncDest***, during a disk synch, indicates that the disk is the destination of a disk synchronization.

If you have replaced a failed disk, follow Procedure 10-12 to resynchronize the replacement drive.

Procedure 10-11xxx
Disabling disk shadowing

Starting point Main menu

1 Select System Status and Maintenance.

2 Select Disk Maintenance.

3 Press the [View Disk Pair] softkey.

You are prompted for a pair number.

4 Enter the number of the pair you want to disable.

5 Press the [Disable] softkey.

You are prompted for the node number.

6 Enter the appropriate node number.

You are prompted to disable primary disk synchronization.

7 Enter **yes** to disable synchronization for the primary disk (this puts the primary disk in the No Access state). Enter **no** to disable synchronization for the shadowed disk (this puts the shadowed disk in the No Access state).

Procedure 10-12xxx
Synching a disk

Starting point Main menu

1 Select System Status and Maintenance.

2 Select Disk Maintenance.

3 Press the [View Disk Pair] softkey.

You are prompted for a pair number.

4 Enter the number of the pair you want to view/synch.

5 Press the [Enable] softkey.

You are prompted for the node number.

If both nodes are InService, you can select either node. If one node is not InService (or InSvStandby), choose the node that is InService.

If you want to speed up the enabling process and both nodes are InService, choose the node that is less busy. Check the DSP Port Status screen (described earlier in this chapter) to check how busy each node is. For MSP nodes, the node that is InSvStandby is always less busy than the InService node.

6 Enter the appropriate node number.

The system determines the source of the synch by choosing the disk that is in ReadWrite mode.

System Event and Error Reports (SEERs)

System Event and Error Reports (SEERs) collect statistics on every system event and error reported by DMS VoiceMail system software components. The reports provide information about the SEER class, SEER number, the date and time that the SEER was generated, and a description of the event or error that occurred at that time.

SEERs are mostly used by maintenance personnel for isolating system faults and repairing hardware and software problems. However, administrators should be able to read, interpret, and assess the severity of events and errors to determine if they are regular events (such as a system audit), errors which can be corrected by the administrator, or if it is necessary to alert support personnel. Once the administrator becomes familiar with SEERs it may also be possible to identify potential problems in their early stages before they become critical errors.

In order to help you judge how serious a system problem might be, SEERs have been classified according to various severity levels. These classifications are based on the impact of the operation that has failed. This reduces the risk of neglecting real problems that have been buried amongst a lot of minor problems or regular system events. When retrieving SEER information, you can therefore filter out all but the most severe problems in order to deal with them quickly.

Each SEER is put into one of the following severity classifications:

- ***Critical* -** indicates any service-affecting problem. A critical problem requires immediate attention, usually from a qualified technician. Examples of critical errors are system reboots, a major base feature not operating, hardware failure (where the system failed to recover from the failure), system capacity reduced below a threshold, software configuration problems, a full volume, a disk drive error.
- ***Major* -** indicates any service-threatening problem. Such problems do not require immediate attention, but will require attention from the administrator or technician to prevent it from becoming a critical problem. A major problem may be allowed to persist up to 24 hours. Examples of major errors are hardware failures from which the system has successfully recovered, unrecovered hardware problems in non-critical components such as tape drives or voice cards, malfunction of a minor feature such as the recording of spoken names or administrative functions, a nearly full volume, a disk drive error (when disks are shadowed), or excessive minor problems.

- ***Minor -*** indicates a problem that has no impact on the system or users of the system. No immediate attention is required on the part of the administrator or a technician. The fault can be allowed to exist for some time. However, an excessive accumulation of minor problems can in itself become a major problem.
- ***Info -*** indicates a normal system event. Knowledge of these events is of use to the administrator as they indicate occurrences such as invalid administrator logon attempts, system time changes, disabled user mailboxes (due to password expiry/violation), successful backups, and the forwarding of non-users to voice messaging.

Each SEER can also be one of several types.

- ***Error -*** Indicates an error which requires the attention of a trained technician.
- ***Admin -*** Indicates an error which can probably be solved by the system administrator. If the administrator is unable to solve the problem, they may call a trained technician.
- ***System -*** Indicates a normal event that should be logged and noted, for example, a successful audit or Operational Measurement collection. This does not sound an alarm.

For a more detailed description of SEERs and their interpretation, see *Maintenance Messages (SEER) Manual* (NTP 297-7001-510).

The System Event and Error Reports screen (Figure 10-13) allows you to set parameters for the type of report you want to generate. In this screen, you are able to specify the range of SEERs that you want included in the report by indicating the class and severity level of the SEERs you wish to monitor. You can also specify the period of time that the report should cover (by entering a start and end date and time). Once the report has been generated according to the criteria you have specified in this screen, you can either view it or print it out.

> ***Note:*** DMS VoiceMail filters SEERs at different levels for printing from the SEER printer. This level can be set so that only those SEERs that the administrator considers important are displayed. SEER filtering is discussed in the *Maintenance Messages (SEER)* manual (NTP 297-7001-510). To reset the SEER filtering level, contact your Regional Support Center (RSC).

Figure 10-13xxx
The System Event and Error Reports screen

```
                    System Status and Maintenance

System Event and Error Reports

SEER Class: 100

Severity Level:  Critical  Major  Minor  [All]
Seer Type:        Error  Admin  System [All}







Report Start(mm/dd/yy hh:mm):05/17/91 04:00  (or blank for oldest)
Report End  (mm/dd/yy hh:mm): ______________  (or blank for newest)

____________________________________________________________________
Select a softkey >

            Cancel      View         Print
                        Reports      Reports
```

The System Event and Error Reports screen contains the following fields:

- ***SEER Class*** **-** This field allows you to specify the class of SEERs that you want to view or print. The SEER class is the code which identifies the type of event or error being reported. There are over 40 classes, each pertaining to a particular software component. Explanations for these codes are given in *Maintenance Messages (SEER) Manual* (NTP 297-7001-510). If you want to retrieve SEERs from all classes, leave the field blank.
- ***Severity Level*** **-** The selection you make in this field determines the SEERs that are displayed in the report by allowing you to selectively view SEERs according to their severity. For a description of the severity levels, see the introduction to this section on SEERs.
  - ***Critical*** retrieves only those SEERs classified as Critical.
  - ***Major*** retrieves those SEERs classified as Major and the level above, namely the Critical Severity SEERs.
  - ***Minor*** retrieves those SEERs classified as Minor and the ones in the levels above, i.e., Major and Critical.
  - ***All*** causes SEERs at all levels of severity to be displayed in the report. This includes the Info level Seers.
- ***SEER Type*** - This field allows you to specify the type of SEERs that you want to view or print.The types are:

- ***Error*** - Error-level SEERs are those which may indicate a system problem, to be corrected by the administrator, possibly with the assistance of technical support. Examples of Error-level SEERs include: hardware errors; software errors; indications that a hardware error may develop.
- ***Admin*** - Administration-level SEERs are those which indicate system problems or configuration difficulties that are likely to be handled by the system administrator without external assistance (for example, a non-DMS VoiceMail user whose calls are forwarded to the DMS VoiceMail system). When the filtering level is set to Admin, the Error-level SEERs are also printed.
- ***System*** - System-level SEERs are those which indicate normal system behaviour, and others which do not require action (for example, nightly audits by the various sub-systems of DMS VoiceMail). When the filtering level is set to System, the Error- and Admin-level SEERs are also printed.
- ***All*** - When All is selected, all SEER types are printed.

• ***Report Start -*** determines the day and time at which the report starts. If this field is left blank, the report starts with the oldest SEER data currently stored in the buffer.

• ***Report End -*** determines the day and time at which the report ends. If this field is left blank, the report will include SEER data up to the last (most recent) entry currently stored in the buffer. If neither the start or end day and time are specified, all SEER data currently stored in the the buffer will be included in the report.

Viewing SEER reports

Once you have filled in the System Event and Error Reports screen, you can either view the report on screen or print it. If you choose to view the report, the screen illustrated in Figure is 10-14 displayed.

Figure 10-14xxx
The Report screen

```
                    System Status and Maintenance

SEER Period from 5/17/91 04:00 to End of SEER data.

Error  Date Time    Type/Severity, Description


  35-0 5/17 04:30  SysInf     SEER registered and active
  26-0 5/17 04:32  SysInf     PP_Base:Number of mappable DSPs on the node
            is 6 0001 0 2030405
  60-0 5/17 04:35  SysInf     Program Completed: VPDMASTER on Node 1
  25-5 5/17 04:50  SysInf     CSL Link is up
  ...       ....   ...        ...
  90-5 5/17 04:50  ErrMin     OCS:Start:Problem Creating New Audit Trail
       File. 1224
  90-5 5/17 04:50  ErrMin     OCS:Audit trail disabled due to a file
       access error, to retry enable in OutCalling Options Screen. 1224

 Select a softkey>
    Exit                                          Next
                                                  Page*
```

*Appears when the information fills more than one screen.

SEER reports contain the following read-only fields:

- ***SEER Period -*** reflects the time period that the report covers. This is determined by the entries that were made in the System Event and Error Reports screen. If no start and end date were entered there, the report will display all SEER data that is currently stored in the buffer.
- ***Error -*** identifies the SEER. The first number indicates the report class (which identifies a particular software component). The second number indicates the report number (which specifies the report within the class, numbered from 0 to 99. This classification system is described in the introduction to the *Maintenance Messages (SEER) Manual* (NTP 297-7001-510). If no class was specified in the System Event and Error Reports screen, SEERs from all classes will be included in the report.
- ***Date & Time -*** indicates the date and time at which the event or error occurred in the system.
- ***Type/Severity -*** indicates the SEER type (Error, Admin or System) and its severity level (Critical, Major, Minor, or Info).

- ***Description -*** gives a brief explanation of the event or the cause for the error.

An alternative method of obtaining SEER information is to monitor the DMS VoiceMail SEER printer, if there is one, thus allowing you to view SEERs as they occur. To do so, SEER printing must be enabled in the General Options screen (it is, by default). Although the format of the report is different from that used by the administration terminal, most of the information is the same (such as the class, number, description, and date and time). In some instances you may also see additional information at the end of the message such as:

RC xxxx

where xxxx is a number signifying a Return Code. These codes provide further information about the SEER and can be found at the back of *Maintenance Messages (SEER) Manual* (NTP 297-7001-510).

Serv. File <filename>

where the filename refers to a voice menu or announcement service ID.

Procedure 10-13xxx
Viewing and printing SEERs

Starting point : The System Status and Maintenance menu, <5> entered.

The System Events and Error Reports screen appears (Figure 10-13).

1. Enter the class of SEERs that you want to retrieve. If you want to retrieve all SEER classes, leave the *Class* field blank.
2. Select a severity level. (To view SEERs at all severity levels, select “ALL”.)
3. Select an error type.
4. If you wish to specify a start and end time for the reporting period, enter the required values in the *Report Start* and *Report End* fields.
5. Choose step 5a to view the report on the terminal, 5b to print the report, or 5c to cancel.
   a. Use [View Reports].

      The report is displayed (Figure 10-14).

      Use [Next Page] to view subsequent pages of the report.

   b. Use [Print Reports].

      You are prompted to make sure your printer is ready and on-line.

      Use [Continue Printing] to continue printing, or use [Cancel Printing] at any time to stop printing. There may be some delay before control is returned to the terminal because the system waits for the printer to stop printing.

   c. Use [Cancel].

      The System Status and Maintenance screen appears.

Operational Measurements

Introduction

The Operational Measurements function gathers data so that you can produce reports containing information about system usage. The following types of information can be collected:

- ***Traffic Data*** - This information is used to monitor the use of system resources, such as DSP ports and disks, and system features, such as voice messaging, voice menu applications, and outcalling. Usage is measured in terms of the number of calls, accesses, or sessions, and their average length. The following traffic reports are available:
  - V oice Services Summary
  - V oice Messaging Detail
  - DSP Port Usage Detail
  - V oice Menus and Announcements Detail
  - Networking Detail (if installed)
  - AMIS Network Detail (if installed)
  - Outcalling Detail (if installed)
  - Disk Usage Detail
- ***User Usage Data*** - This information is used to monitor how specific users are making use of voice messaging, Meridian networking and AMIS networking (if installed). Data is broken down to show activity on a daily basis. User usage reports display the following information about each user:
  - the number of times a user has logged on
  - the number of times callers have connected to a user's mailbox through the express messaging and call answering services
  - the total connect time for all user logons, express messaging and call answering sessions
  - the number of messages created during logon
  - the number of messages received through the express messaging and call answering services

- the total message length (for all messages created and received by a user)
- the disk space used by the user's messages and greetings

- ***Billing Data*** - This information is used to monitor how much each user uses the various voice messaging features. This information can then be used for billing purposes. This category is further divided into local billing data and networking billing data. (Networking data includes both AMIS and Meridian networking data. Meridian networking is a proprietary networking protocol which may be used with CPE systems.)

  DMS VoiceMail only collects this data. To use the data to generate billing reports, you must use AdminPlus, an optional software program. See the *AdminPlus System Administration Guide*. You can also use AdminPlus to view billing information on a per session basis (known as a "session trace"). This is documented in the "File Download" chapter under the section heading "Billing Files".

This data can be collected over time and at intervals specified by you. The information can be viewed or printed as Operational Measurement reports. These reports are discussed in the sections that follow.

When to use Operational Measurements

The need to use Operational Measurements is influenced by many factors. Your particular situation may dictate that you monitor your system on a frequent basis or perhaps not at all. Operational Measurements can be made in the following three categories:

- As an accounting and billing tool, Operational Measurements is used to generate the daily user billing files (for either local or networking activity) that you subsequently download to AdminPlus (see the *AdminPlus System Administration Guide*). (Note that AdminPlus is only available for CPE systems.) If your organization does not bill users of DMS VoiceMail, you may not need to use the User Usage component of Operational Measurements. It however can also be used for tracking problems/history or for security reasons (e.g., who called whom, when, and for how long).
- As a capacity planning tool, Operational Measurements is used to generate traffic reports that you subsequently analyze to determine whether your system requires an upgrade either in disk storage, channel capacity, or perhaps in the number of nodes (should the number of users on your system approach one of the limits discussed in *The DMS VoiceMail Product Guide* (NTP 297-7001-010)). If your organization's use of DMS VoiceMail is fairly stable, you need only use the traffic measurement component of Operational Measurements on an infrequent basis to verify that the system's resources are adequate for your needs.
- As a monitoring tool, Operational Measurements is used to generate reports on the use of various system and user-defined features. If your site makes extensive use of voice services, especially voice menu applications, you may need to monitor the use of these services to determine which services are used most frequently, and to assist you in providing a high quality of service to your users.

Once you decide on which aspects of Operational Measurements you require, you then need to determine how to collect and process this information.

Traffic Measurements

- The beginning and end of the traffic reporting period. You need not monitor traffic on a continuous basis; perhaps you require traffic data only at peak hours in order to assess whether your system meets peak load requirements.
- The number of intervals within the reporting period. The number of intervals is determined by how fine an analysis you wish to perform during the reporting period. For coarse measurements, you would choose to divide the reporting period into only a few intervals; for finer measurements, you would divide it into several intervals.
- The number of traffic periods held on the system at any one time.

User Usage Information

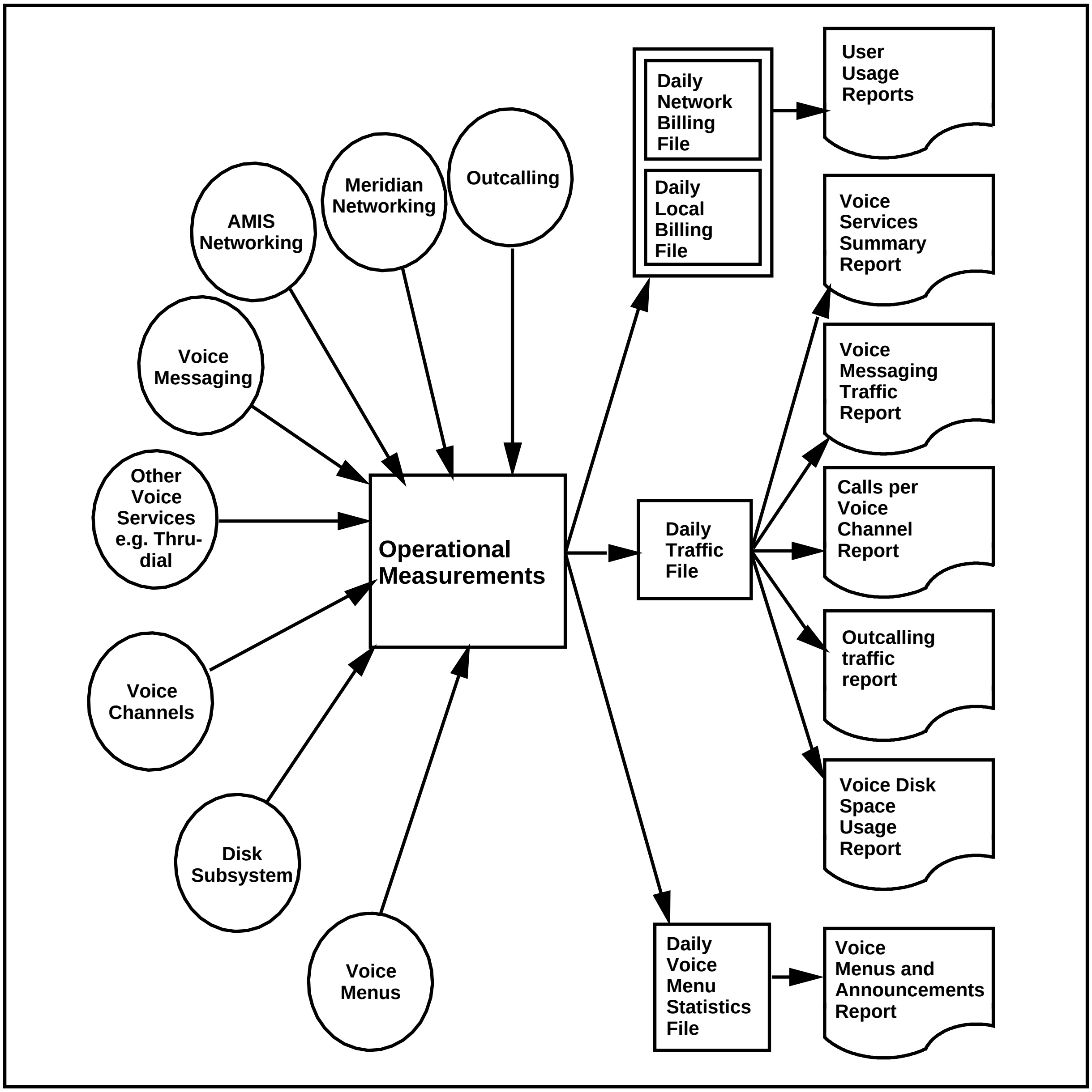
- The number of user usage days the data is to be held on the system at any one time.

Because Operational Measurements are kept on hard disk, they are periodically overwritten (as determined by the number of days you specified to be kept on disk), and it is important that you view or print these reports before the system overwrites them with new information.

Overview of Operational Measurements

Operational Measurements screen a comprehensive record of system and user activity. When a user accesses a service provided by DMS VoiceMail, the user's actions are recorded in several ways, and may be reflected in several Operational Measurement reports (see Figure 11-1).

Figure 11-1xxx
Operational measurements and reports



Example

The following scenario presents typical interactions with the system and describes the impact these interactions have on Operational Measurements. In order to follow this example in detail, refer to the descriptions of the reports in this section. The following three events are assumed to have taken place within 30 minutes (one reporting interval) of each other:

- A user logged on twice to a mailbox (DMS VoiceMail Voice Messaging service) to compose, play, and reply to messages. The first call lasted 289 seconds, the second lasted 61 seconds.
- The user then placed two calls to a voice service defined by the system administrator. The first call lasted 73 seconds and involved interaction with the main menu of the service. The second call lasted 29 seconds and involved interaction with the main menu of the service and one sub-menu. The sub-menu was used to access Voice Messaging for 160 seconds.
- The user received an Express Message; the call to place this message (made by another user) lasted 45 seconds.

Voice Service Summary

This report lists activity for each service provided by DMS VoiceMail. The sample report on the following page shows that two calls were placed to the Voice Menu service for a total of 102 seconds (73 + 29), or an average of 51 seconds per half-hour interval. The CCS count (hundred call-seconds) is computed for the half-hour interval (2 intervals per hour) as follows:

$$\text{CCS} = \frac{\text{102 seconds x 2 intervals}}{\text{100 seconds}} = 2.04 \ \ (2)$$

Similarly, three calls were placed to Voice Messaging, two directly and one through the Voice Menu service, for a total duration of 510 seconds (289 + 61 + 160), or an average length of 170 seconds. The CCS is computed for the half-hour interval as follows:

$$\text{CCS} = \frac{\text{510 seconds x 2 intervals}}{\text{100 seconds}} = 10.2 \ \ (10)$$

One call was placed to Express Messaging, for a total duration of 45 seconds, or an average length of 45 seconds. The CCS is computed for the half hour interval as follows:

$$\text{CCS} = \frac{\text{45 seconds x 2 intervals}}{\text{100 seconds}} = 0.90 \ \ (1)$$

Voice Service Summary

Interval	Start-End	Service Name	Number of Accesses	Average Length (in seconds)	Voice Mail Usage (in CCS)
8/14	16:30-17:00	Thru-dial	0	0	0
8/14	16:30-17:00	Voice Menu	2	51	1
8/14	16:30-17:00	Voice Messaging	3	170	3
8/14	16:30-17:00	Call Answering	0	0	0
8/14	16:30-17:00	Express Messaging	1	45	1

Voice Messaging Detail

Four internal calls were recorded; three calls were logon sessions, and one call was an Express Messaging session. Messages were created during this interval: one message during the Express Messaging session and two messages were composed. Therefore, two out of the three times that the user logged on to DMS VoiceMail, a message was composed. Message length statistics apply to all messages created, including the Express Messaging message. Hold time statistics apply to the three logon sessions and the Express Messaging session.

Voice Messaging (VM Logon, Call Answering, and Express Messaging)

		Number of Calls		Number of Sessions		Session Length		Messages Created		Message Length	
Interval	Start-End	Int	Ext	EM/Ans	Log	Avg	Max	EM/Ans	Log	Avg	Max
8/14	16:30-17:00	4	0	1	3	51	289	1	2	16	51

Voice Menus and Announcements

This report shows that there were two direct accesses to voice menu 1300 and one indirect access to voice menu 1301 (accessed through menu 1300). Announcement 300 was accessed twice.

Voice Menus and Announcement Detail

Interval / ID	Start-End / Service Accesses	For each menu item, the number of accesses are: 1	2	3	4	5	6	7	8	9	0	*	#
8/14	16:30-17:00												
M 1300	2	1	1	1	1	1	0	0	0	0	0	4	0
M 1301	1	0	1	1	0	0	0	0	0	0	0	0	0
A 300	2	0	0	0	0	0	0	0	0	0	0	0	0

Meridian Networking Detail

This report displays Meridian networking activity and it only applies to CPE systems with Meridian Networking installed. During the interval period, the host received a total of 100 messages from site 111, and 25 messages from site 112. The host delivered a total of 25 messages to site 111, and 21 messages to site 112. One message was not delivered to site 112 within the stale-dating threshold. Two attempted calls were made to site 111; one was successful, and the time used by the networking calls totalled 2:32. Six messages did not reach both site 111 and site 112 due to a lack of available resources, nine messages did not reach site 111 because the remote site could not be accessed, and ten calls did not reach site 112 due to protocol errors.

```
Networking Detail

Interval Start-End
     Messages          Messages Delivered                                     Failures
     Received      Eco  Std  Urg  NDN  Ack    Failed      Network Usage  No  Not   Prot
Site (from site)  (--------to site-------)  to Send   Att  Suc  Time  Res Reach Error
                                                                (min)

8/14      10:00-10:30
111  100           10  10   5     0    0    0           2   1   2:32  6     9   0
112   25           10  8   3      0    0    1           0   0   0:00  6     0   10
```

DSP Port Usage Detail

A channel is allocated to each call placed to DMS VoiceMail. This report indicates which of the eight channels were used to handle the five calls.

```
DSP Port Usage Detail

                              Number of  Number of  Incoming    Outgoing    Voice Mail
                              Incoming   Outgoing   Avg Length  Avg Length   Usage
Interval Start-End    Channel  Calls     Calls   (in seconds)  (in seconds) (in CCS)

08/14    16:30-17:00      1       0       0            0          0            0
08/14    16:30-17:00      2       1       0          290          0            6
08/14    16:30-17:00      3       2       0           58          0            2
08/14    16:30-17:00      4       1       0          190          0            4
08/14    16:30-17:00      5       0       0            0          0            0
08/14    16:30-17:00      6       0       0            0          0            0
08/14    16:30-17:00      7       0       0            0          0            0
08/14    16:30-17:00      8       1       0           61          0            1
```

For channel 2, the CCS calculations are performed as follows:

$$\text{CCS} = \frac{\text{290 seconds x 2 intervals}}{\text{100 seconds}} = 5.80 \ (6)$$

For channel 3, the CCS calculations are performed as follows (keeping in mind that the two calls lasted 45 and 73 seconds):

$$\text{CCS} = \frac{\text{(45 seconds + 73 seconds) x 2 intervals}}{\text{100 seconds}} = 2.36\ (2)$$

For channel 4, the CCS calculations are performed as follows:

$$\text{CCS} = \frac{\text{190 seconds x 2 intervals}}{\text{100 seconds}} = 3.80\ \ (4)$$

For channel 8, the CCS calculations are performed as follows:

$$\text{CCS} = \frac{\text{61 seconds x 2 intervals}}{\text{100 seconds}} = 1.22\ \ (1)$$

Disk Usage Detail

This report displays the voice storage used in hours and minutes, and as a percentage of the total voice storage available per volume.

```
Disk Usage Detail

                              Volume  Voice Volume    Voice Space   Text Space
Interval   Start-End           Name   Size (hh:mm)     Used (%)      Used (%)

08/14     16:30-17:00         VS1        1:51             18            18
                               VS2        5:53             54            54
```

T1 Link Handler Detail

This report displays the number of errors encountered on the T1 links during a given reporting interval.

```
T1 Link Handler Detail

                                                         Out of                  Backward    Forward
                                            Bipolar      Frame  Extended        Slip        Slip
Interval   Start-End     T1 Link    Violations   Errors SF Errors        Count       Count

01/01/92   09:00-10:00 13-1-1         0             0        0             0           0
01/01/92   09:00-10:00 13-1-2         0             0        0             0           0
01/01/92   09:00-10:00 13-1-3         0             0        0             0           0
01/01/92   09:00-10:00 13-1-4         0             0        0             0           0
01/01/92   09:00-10:00 14-1-1         0             0        0             0           0
01/01/92   09:00-10:00 14-1-2         0             0        0             0           0
01/01/92   09:00-10:00 14-1-3         0             0        0             0           0
01/01/92   09:00-10:00 14-1-4         0             0        0             0           0
```

Voice Messaging User Usage Report

Local Usage: This report shows that user Smith had a total of 18 call answering and/or express messaging sessions, and a total of seven logon sessions over the two days shown in the report. The value that appears under Number of Messages, EM/Ans, refers to the total number of messages from Call Answering, the number of messages composed, forwarded, and replied to.

```
Last Name    First Name     Department     Mailbox      Billing Class

Smith        David          T20            2255          1

Local Usage:
         Number of      Connect     Number of      Message      Disk
          Sessions       Time       Messages       Length       Used
Date     EM/Ans  Logon  (mm:ss)   EM/Ans  Logon    (mm:ss)     (mm:ss)
02/12/90   10      4      4:00      9      2         6:30        4:30
02/13/90    8      3      3:12      8      3        12:35        5:30
---------------------------------------------------------------------
Total      18      7      7:12     17      5        19:05
```

Meridian Networking Usage: This report shows that user Smith sent a total of 20 economy messages, 17 standard messages, and 6 urgent messages over the two days shown in this report. The total length of the messages appears for each message type.

```
Last Name    First Name     Department     Mailbox      Billing Class

Smith        David          T20            2255          1

Meridian Networking Usage:

         Number of    Total     Number of    Total   Number of    Total
         Economy      Length    Standard     Length  Urgent       Length
Date     Messages     (mm:ss)   Messages     (mm:ss) Messages     (mm:ss)
02/12/90    12          4:12      10          2:30     6           4:10
02/13/90     8          2:23       7         11:40     0           0:00
         --------------------  ------------------  ------------------
            20          6:35      17         14:10     6
```

AMIS Network Usage: This report shows that user Smith sent a total of 20 economy messages, 8 standard messages, and no urgent messages over the two days shown in this report. The total length of the messages appears for each message type.

```
Last Name    First Name     Department     Mailbox     Billing Class

Smith        David          T20            2255          1

AMIS Network Usage:

          Number of    Total     Number of    Total   Number of    Total
          Economy      Length    Standard     Length  Urgent       Length
Date      Messages     (mm:ss)   Messages     (mm:ss) Messages     (mm:ss)
02/12/90    10           3:10       1           1:30    0            0:00
02/13/90    10           1:20       7           5:10    0            0:00
          --------------------  ------------------  ------------------
            20           4:30       8           6:10    0            0:00
```

Disk capacity

In order to calculate your projected storage requirements, you must determine the reporting intervals and the number of reporting periods you wish to store on disk for each of the three types of operational measurements (see “Operational Measurement Options” on page 11-14 for details).

Because operational measurement data must be stored in a finite amount of disk space, it is periodically overwritten by new data. You must ensure that you view or print any vital information before it is overwritten. You must also ensure that operational measurement data does not exceed the available storage capacity.

The amount of storage required for each operational measurement can be estimated from Table 11-1. Once you have this information, compute the storage as follows:

If User Usage is enabled, 2 days of billing data will be stored.

Total storage	=	2 x Billing Data Cost
		+ number of traffic days x 1%
		+ number of user usage days x cost of user usage days

Example: 20 channel system with 1000 users, 31 user usage days, 4 days of traffic stored.

Total storage = (2x8%) + (4x1%) + (0.7x31) = 42%

The total storage cannot exceed 100%, or you are likely to run out of disk space. Should your calculations yield a result greater than 100%, reduce one or all of the stored billing periods, and repeat your calculations. The values presented in Table 11-1 are based on typical parameters for various DMS VoiceMail configurations. Should your system deviate markedly in any of these assumed traffic patterns, you will need to experiment to determine what your system can accommodate.

Table 11-1xxx
Storage requirements for operational measurements

System type	Number of users	Billing Data Cost	User Usage Data Cost
12 channel, 5 hr	500 1000	5% per day 10% per day	0.4% per day 1.0% per day
12 channel, 24, 36, 54 hr	1000	5% per day	0.7% per day
20 channel, 26, 54, 84, 114 hrs	1000 2000 3500	8% per day 10% per day 21% per day	0.7% per day 1.4% per day 2.2% per day
36 channel, 45, 90, 120, 180 hrs	3750 7500	10% per day 18% per day	1.0% per day 2.0% per day
48 channel	5000 10000	12% per day 25% per day	1.0% per day 2.0% per day

Assumptions:

System in use 12 hours per day, 5 days per week, at an average of 0.5 peak traffic
Average holding time is 40 seconds.
Voice menu traffic is 10% of voice message traffic
Operational Measurements set for one hour commit interval

The Operational Measurements menu

The items listed in the Operational Measurements menu (Figure 11-2) allow you to access screens that are used to set parameters related to the collection and storage of data and to view and print traffic reports and user usage reports.

Figure 11-2xxx
The Operational Measurements menu

```
                      Operational Measurements


          1 Operational Measurement Options

          2 Traffic Reports

          3 User Usage Reports







 ____________________________________________________________
 Select an item >

   Exit
```

Procedure 11-1xxx
Using the Operational Measurements menu

Starting point : The Main Menu, <6> entered.

The Operational Measurements menu appears (Figure 11-2).

1 To choose an item enter its number and press <Return>.

The menu corresponding to your selection appears. See the following sections later in this chapter for details:

<1> "Operational Measurement Options" (collection parameters)

<2> "Traffic Reports" (for viewing and printing reports)

<3> "User Usage Reports" (for viewing and printing reports)

2 Use [Exit] to return to the Main Menu.

Operational Measurement Options

Operational Measurement Options (Figure 11-3) define parameters for the collection of system and user statistics. This includes the time at which traffic data collection begins and ends every day, how often collected traffic statistics are written to disk, whether nor not user usage data is collected, and if so, the number of days that user usage data is stored before being deleted.

Figure 11-3xxx
The Operational Measurement Options screen

```
                    Operational Measurements

Operational Measurement Options

Collect Traffic Data:                     Disabled  [Enabled]

Traffic Period Start (hh:mm)              01:00

Traffic Period End (hh:mm):               01:00

Traffic Commit Interval (hh:mm):          01:00

Number of days of Traffic Data stored:     8

Collect User Usage Data:                  [Disabled]  Enabled

Number of days of User Usage Data stored:  31

    Save       Cancel
```

The following fields are displayed:

- ***Collect Traffic Data*** - When this field is “Enabled” a statistical record of voice messaging and other voice services, voice channel traffic, networking message traffic (AMIS or Meridian Networking), and voice disk space usage will be collected and stored on disk. The default is “Enabled”.
- ***Traffic Period Start (hh:mm)*** - The time at which data begins to be collected, based on the 24-hour clock. The valid range is from 00:00 to 23:30. You may only enter values in half-hour increments, for example, 01:00, 01:30; 02:00, 02:30, etc. The default is “01:00”.

- ***Traffic Period End (hh:mm)*** - The time at which data stops being collected, based on the 24-hour clock. The valid range is from 00:00 to 23:30. You may only enter values in half-hour increments, for example, 01:00, 01:30; 02:00, 02:30, etc. To continuously collect traffic data, set the Period Start equal to the Period End (i.e., Period Start = 01:00 and Period End = 01:00). In this manner, data will be collected 24 hours a day. The default is “01:00”.
- ***Traffic Commit Interval (hh:mm)*** - The value entered in this field determines how often the collected traffic statistics are written to the hard disk within the defined traffic period. The default is “01:00”. The valid range is from 00:00 to 23:30. For example, if *Collect Traffic Data* is set to “Enabled” and

  Traffic Period Start = 8:00 am,
  Traffic Period Stop = 4:00 pm,
  Traffic Commit Interval = 30 minutes,

  traffic data is collected between 8:00 a.m. and 4:00 p.m. daily and traffic reports are written to the hard disk every 30 minutes during this period. The first report is written out at 8:30 a.m. and the last one is written out at 4:00 p.m. The traffic commit interval default setting is “01:00”. Commit intervals should be entered in half-hour increments and equally divisible into the period range.

  Note: The traffic commit interval can be set to 24 hours, however an interval greater than 2 hours is not recommended because the accumulated numbers may be too large to be accommodated by the fields in the report screens. If this is the case, a series of asterisks are displayed in the field to indicate overflow. Furthermore, any data that is not committed to disk is lost if a system reboot occurs.
- ***Number of days of Traffic Data Stored*** - This field determines the number of days that traffic data is maintained before being overwritten by new traffic data. For example, if this field is set to 8, on the 9th day you will not be able to view traffic data collected on the first day as it will have been overwritten, but you will be able to view the data from the remaining eight days. The valid range is from 1 to 8 days. The default is “8”.
- ***Collect User Usage Data*** - This field controls the collection of daily user usage data and may be “Disabled” or “Enabled”.
- ***Number of days of User Usage Data Stored*** - This field determines the number of days that information about user activity is kept on the hard disk. The range is from 1 to 60. The default is “31”.

Traffic Reports

The Traffic Reports screen appears when item <2> is selected from the Operational Measurements menu.

Figure 11-4xxx
The Traffic Reports screen

```
                 Operational Measurements
Traffic Reports

Voice Service Summary:                  [No] Yes
Voice Messaging Detail:                 [No] Yes
DSP Port Usage Detail:                  [No] Yes
Voice Menus and Announcements Detail:[No] Yes
Networking Detail*:                     [No] Yes
AMIS Networking Detail:                 [No] Yes
Outcalling Detail:                      [No] Yes
Disk Usage Detail:                      [No] Yes
T1 Link Handler Detail:                 [No] Yes


Report Start(dd/mm/yy hh:mm):____________________ (or blank for oldest)
Report End  (dd/mm/yy hh:mm):____________________ (or blank for newest)

              Cancel      View        Print
                          Reports     Reports
```

* Meridian Networking is an optional feature for CPE systems only.

The following fields are displayed in the Traffic Reports screen:

- ***Voice Services Summary*** - displays statistics for the services that are installed on system.
- ***Voice Messaging Detail*** - displays statistics for voice messaging usage. This includes information about the number of messages created in various categories, average message lengths, hold times, and the number of internal and external calls.
- ***DSP Port Usage Detail*** - displays statistics, including the number of incoming and outgoing calls, for each DSP port/channel.
- ***Voice Menus and Announcements Detail*** - displays statistics for voice menu and announcements services. The report displays the number of times that each voice menu option, in the specified voice menu application, has been used within the specified reporting period. This only shows if the feature is installed.

- ***Meridian Networking Detail*** - This field appears only if Meridian Networking (applicable only to CPE systems) is installed. This report displays information about the number of economy, standard, urgent, non-delivery notification, and acknowledged networking messages sent and received at the remote sites, as well as connection statistics.
- ***AMIS Networking Detail*** - This field appears only if AMIS Networking is installed. This report displays information about the number of non-delivery notifications (NDNs), economy, standard and urgent messages sent and received by the system, as well as connection statistics.
- ***Outcalling Detail*** - This field appears only if Outcalling is installed. The Outcalling Detail report displays statistics for Remote Notification and Delivery to Non-Users activity.
- ***Disk Usage Detail*** - This report summarizes how much voice space and text space have been used for each voice storage volume.
- ***Report Start (dd/mm/yy hh:mm)*** - When requesting reports, this field allows you to specify the date and time at which the report should begin. The value you enter is based on the 24-hour clock. The valid range is from 00:00 to 23:59 (12:00 midnight to 11:59 p.m.). If this field is left blank, the default-the start of available data-is used.
- ***T1 Link Handler Detail -*** This reports shows the error counts for the T1 links during a given reporting interval.
- ***Report End (dd/mm/yy hh:mm)*** - This field determines the date and time at which the report should end. The value entered in this field, based on the 24-hour clock, can be set from 00:00 to 23:59. If this field is left blank, the default-the end of the available data-is used.

Procedure 11-2xxx
Viewing and printing Traffic Reports

Starting point : The Operational Measurements screen, <2> entered.

1 The Traffic Reports screen appears (Figure 11-4).

2 Select the reports you wish to view.

3 (This step is optional.) Specify start and stop times for the report period by entering the values in the *Report Start* and *Report End* fields.

4 Choose step 4a to view the reports on the terminal, 4b to print the reports, or 4c to cancel.

a. Use [View Reports].

The selected report screens are displayed (see the following pages for descriptions of each report).

When you select the various reports screens, you will see <Exit Current> and <Exit All> softkeys at the bottom of the screen. <Exit Current> lets you exit from the current report screen to the next report screen,while <Exit All> lets you exit from all the reports screens back to the Traffic Reports screen.

Use [Next Page] to view subsequent pages of the current report.

b. Use [Print Reports].

You are prompted to ensure the printer is ready and on-line.

Use [Continue Printing] to print the reports, or use [Cancel Printing] at any time to cancel printing (there may be some delay before control is returned to the screen because it waits for the printer to stop printing).

c. Use [Cancel].

The Operational Measurements menu is redisplayed.

Voice Service Summary Report

The Voice Service Summary Report provides statistics for each of the voice services installed in your system (Figure 11-5). The total number of direct accesses to a service, that is the number of times a user dialed the service directly, and the average length of each access are given. Indirect accesses, through other services such as voice menus, are not displayed in this report.

Figure 11-5xxx
Voice Service Summary Report screen

```
                        Operational Measurements
Voice Service Summary
                                                              Average   Voice
                                          Number of    Length    Mail Usage
Interval Start-End       Service Name     Accesses  (in seconds)(in CCS)

04/22     09:00-10:00    Thru-Dial             5         60          3
04/22     09:00-10:00    Voice Menus          10         30          3
04/22     09:00-10:00    Voice Messaging      10         30          3
04/22     09:00-10:00    Call Answering       60         30         18
04/22     09:00-10:00    AMIS                 12         50          6
04/22     09:00-10:00    Express Messaging    10         60          6
04/22     09:00-10:00    Voice Announcements   5         60          3
04/22     09:00-10:00    Networking *         10         60          6
04/22     09:00-10:00    Voice Administration  0          0          0
04/22     09:00-10:00    Voice Prompt Admin    0          0          0
04/22     09:00-10:00    Time of Day Control   0          0          0
04/22     09:00-10:00    Delivery to Non User  5          0          0
04/22     09:00-10:00    Remote Notification   0          0          3
04/22     09:00-10:00    Remote Activation     0          0          0
04/22     09:00-10:00    Voice Forms          60         25          4
04/22     09:00-10:00    Transcription Service 40        20          2

 Select a softkey >

 Exit        Exit                    Next
 Current     All                     Page**
```

* Available as an option only on CPE systems.
**Appears when the information fills more than one screen.

The following read-only fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***Service Name*** - displays the name of the service that was accessed.
- ***Number of Accesses*** - displays the number of direct calls made to the corresponding service.
- ***Average Length (seconds)*** - displays the average length of the corresponding voice service sessions during the specified interval.
- ***VoiceMail Usage (in CCS)*** - displays the amount of time that a DMS VoiceMail service was active in the defined interval. The value is given in CCS (hundred call-seconds), a traffic measurement statistic. One CCS is equal to 100 seconds of call connection time.

Voice Messaging Detail Report

The Voice Messaging Detail Report (Figure 11-6) provides information about logon sessions, call answering sessions, and messages composed during logon sessions. If data is unavailable for a given statistic, “N/A” (not available) is displayed instead of a value; if the value exceeds the capacity of the display, “>999” is displayed.

Figure 11-6xxx
The Voice Messaging Detail Report

Operational Measurements

Voice Messaging Detail

Interval Start-End	Number of Calls Int	Ext	Number of Sessions EM/Ans	Log	Session Length Avg	Max	Messages Created EM/Ans	Log	Message Length Avg	Max
10/28 09:00-10:00	18	0	2	16	238	470	0	2	20	25
10/28 10:00-11:00	12	2	5	9	310	310	1	32	14	14
10/28 11:00-12:00	17	1	1	17	478	624	1	20	20	20

Select a softkey >

Exit Current	Exit All		Next Page*	

*Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***Number of Calls*** - The number of calls made to DMS V oiceMail. More specifically,

- ***Int*** indicates the number of calls made from inside the switch during the specified interval.
- ***Ext*** indicates the number of calls made from outside the switch during the specified interval.

- ***Number of Sessions (EM/Ans and Log)*** - The number of express messaging, call answering and logon sessions that occurred during the interval. The sum of the values in these two columns should equal the sum of the two *Number of Calls* values.To determine the number of messages that were actually received or created during these sessions, check the *Messages Created* field.
- ***Session Length (Avg and Max)*** - The average length and maximum length (in seconds) of call answering and logon sessions for the interval.
- ***Messages Created*** - The number of messages created during the interval.
  - ***EM/Ans*** indicates the number of messages left during express messaging and call answering services.
  - ***Log*** indicates the number of messages that were created (using the compose, forward or reply command) during the interval.
- ***Message Length (Avg and Max)*** - The average length and maximum length (in seconds) of messages received and created during the interval. Since message length impacts disk storage, use this information to determine if enough disk space has been provisioned for voice messages.

DSP Port Usage Detail Report

DSP Port Usage Detail Reports detail channel activity for incoming and outgoing calls, including average session lengths for each type as well as CCS (hundred call-seconds) statistics.

Figure 11-7xxx
DSP Port Usage Detail Report

Operational Measurements

DSP Port Usage Detail

Interval	Start-End	Channel	Number of Incoming Calls	Number of Outgoing Calls	Incoming Avg Length (in seconds)	Outgoing Avg Length (in seconds)	Voice Mail Usage (in CCS)
04/22	09:00-10:00	1	10	0	120	0	12.2
04/22	09:00-10:00	2	5	0	60	0	12.0
04/22	09:00-10:00	3	1	1	96	0	7.5
04/22	09:00-10:00	4	1	0	30	0	3.4
04/22	09:00-10:00	5	2	1	57	0	4.8
04/22	09:00-10:00	6	0	0	0	0	0.0
04/22	09:00-10:00	7	0	0	0	0	0.0
04/22	09:00-10:00	8	3	4	47	98	11.1
04/22	09:00-10:00	9	5	0	90	0	8.0
04/22	09:00-10:00	10	5	0	90	0	8.2
04/22	09:00-10:00	11	5	0	90	0	8.7
04/22	09:00-10:00	12	5	0	90	0	8.0

Select a softkey >

Exit Current	Exit All		Next Page*	

*Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***Channel*** - the channel being monitored.

- ***Number of Incoming Calls*** - Number of calls incoming during the interval.
- ***Number of Outgoing Calls*** - Number of calls outgoing during the interval. (This value is 0 for services which do not use thru-dialers.)
- ***Incoming Avg Length (seconds***) - Average length of incoming calls during the interval.
- ***Outgoing Avg Length (seconds)*** - Average length of outgoing calls during the interval.
- ***Voice Mail Usage (CCS)*** - Represents the amount of time in terms of CCS (hundred call-seconds) that a DMS VoiceMail channel was active in the defined interval. CCS is a traffic measurement statistic. One CCS is equal to 100 seconds of call connection time. The value is displayed in the nearest one tenth of a CCS (for example, 11.0).

  Note: There is a similar *Voice Mail Usage* field in the Voice Services Summary screen (Figure 11-5). However, because the two fields measure usage differently (one in terms of channels and the other in terms of voice services), there may be small differences between the two fields if you calculate the totals for the displayed values.

Voice Menus and Announcements Detail report

The Voice Menus and Announcements Detail report records the number of times that each menu option in a voice menu application was used during the reporting period. This report details all accesses, direct or indirect, to voice menus and announcements. Direct accesses occur when a user dials the DN of the menu or announcement. Indirect accesses occur when a service is accessed from another service through a menu selection.

Figure 11-8xxx
The Voice Menus and Announcements Detail Report

```
                        Operational Measurements

Voice Menus and Announcements Detail


Interval  Start-End
          Service     For each menu item, the number of accesses are:
  ID      Accesses       1   2   3   4   5   6   7   8   9   0   * #

9/30      09:00-10:00    No Activity
9/30      10:00-11:00
M 1234      9            0   0   1   2   0   1   0   0   0   1   1 0
M 1235     10            2   1   0   1   0   0   0   0   0   1   5 0
A 1237      2            0   0   0   0   0   0   0   0   0   0   0 0
9/30      11:00-12:00
M 1234      1            1   0   0   0   0   0   0   0   0   0   0 0
A 1300      4            0   0   0   0   0   0   0   0   0   0   0 0
9/30      12:00-13:00    No Activity
9/30      13:00-14:00
A 1305      3            0   0   0   0   0   0   0   0   0   0   0 0
M 1239      9            1   3   2   0   0   0   1   0   1   0   2 0
________________________________________________________________________
Select a softkey >
   Exit          Exit                          Next Page*
  Current        All
```

*Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.

- ***ID*** - This is the ID number of the voice menu or announcement. V oice menus are indicated by the letter “M” followed by the ID number. Announcements are preceded by an “A”.
- ***Service Accesses*** **-** The number of times the menu or announcement was accessed (either directly or indirectly) during the measurement period.
- ***For Each Menu Item, the Number of Access Are*** **-** The total number of times that each menu option was used during the measurement period. For announcements, all frequencies are “0” since announcements do not process digits.

Networking Detail Report

This report (Figure 11-9) displays traffic totals for each site within the DMS VoiceMail network with Meridian Networking installed. This is only applicable to CPE systems. Statistics are shown for the number of messages received at each site from other network sites and the messages delivered to network sites. Statistics are also displayed for network usage and failures.

Figure 11-9xxx
The Networking Detail Report

```
                              Operational Measurements


 Networking Detail

 Interval Start-End
        Messages       Messages Delivered                                       Failures
        Received       Eco  Std  Urg  NDN  Ack  Failed Network Usage  No  Not   Prot
 Site   (from site)  (--------to site--------) to sendAtt  Suc Time  Res Reach Error
                                                                (min)

 09/30  10;00-11:00
 111     100            10  10   5    0    0    0     2    1    2:32 6   12     0
 112     100             0   0    2   0    0    0     4    2    4:04 0    0     0
 09/30  11:00-12:00
 112     25             10  10   5    0    0    0     0    0    0:00 6    0    12

 ______________________________________________________________________________

 Select a softkey >

   Exit            Exit                             Next Page*
  Current          All
```

* Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.

- ***Site*** - The site ID.
- ***Messages Received*** - The number of messages received by the identified site from other networking sites.
- ***Messages Delivered*** - The number of messages delivered from the identified site to remote sites in the network . This statistic is further subdivided into the following categories based on the type of message:
  - ***Eco*** - These are messages that have been classed as economy.
  - ***Std*** - These are messages that have been classed as standard.
  - ***Urg*** - These are messages that have been classed as urgent.
  - ***NDN*** - (Non-delivery Notification) These are messages sent by the system to the users whose messages could not be delivered (as a result of a "Failed to send".
  - ***Ack*** - Acknowledgements are returned by Networking to indicate that a message (that was tagged for acknowledgement) was read by the DMS VoiceMail user at the remote site.
  - ***Failed to Send*** - These are messages that have not been delivered to the remote site within the stale-dating threshold (see the "Meridian Networking" chapter for information about stale-dating). They are effectively "lost" and must be re-composed and re-sent.
- ***Network Usage*** - This statistic indicates the number of networking calls placed by the site during the specified interval. It is further broken down into the following categories:
  - ***Att*** - indicates the number of attempted calls.
  - ***Suc*** - indicates the number of successful calls.
  - ***Time*** - indicates the total amount of time (in minutes) used by networking calls.
- ***Failures*** - A failure refers to a single unsuccessful attempt to send a networking message. Networking will attempt to send these messages the next time it is scheduled to send messages to the remote site. If a message experiences many failures, and is not delivered within a certain period of time, it will be reported in the *Failed to Send* field.
  - ***No Res*** - or *no resources*, means that the modems or voice ports could not be accessed by Networking to send these messages to the remote site.
  - ***Not Reach*** - or *not reachable*, means that the remote site could not be accessed.
  - ***Prot Error*** - or *protocol error*, means that the connection was made to the remote site, but message delivery was prevented by a protocol error.

AMIS Detail

This screen is displayed only on those systems with AMIS networking capability. The AMIS Detail report (Figure 11-10) displays traffic totals for your site. Statistics are shown for the number of AMIS messages received at your site and delivered to other sites, the connect time, and the number of failures for each time interval displayed in the report.

Figure 11-10xxx
The AMIS Detail report

Operational Measurements

AMIS Detail

Interval	Start-End	Messages Received	Messages Delivered Eco	Std	Urg	NDN	Failed	Connect Time (mm:ss)	Failures No Res	Not Reach	Prot Error
09/30	10:00-11:00	12	0	5	2	0	0	4:00	0	1	1
09/30	11:00-12:00	0	0	2	0	0	0	2:00	0	1	0
09/30	12:00-13:00	24	0	5	1	0	0	8:00	0	0	1
09/30	13:00-14:00	6	0	2	1	0	0	3:00	0	1	1

Select a softkey >

Exit Current	Exit All		Next Page*	

* Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.

- ***Messages Received -*** indicates the number of AMIS messages that were received at the local site during the time interval indicated.
- ***Messages Delivered -*** indicates the number of AMIS messages (originating from the local site) that were delivered to other voice messaging systems during the interval indicated. This statistic is further subdivided according to the type of message.
  - ***Eco*** - The number of messages, tagged as economy, that were delivered to other AMIS sites during the specified interval.
  - ***Std*** - The number of messages, tagged as standard, that were delivered to other AMIS sites during the specified interval.
  - ***Urg*** - The number of messages, tagged as urgent, that were delivered to other AMIS sites during the specified interval.
  - ***NDN*** - (Non-delivery Notification) The number of NDN messages sent by the system during the specified interval.
  - ***Failed*** - The number of unsent messages. These messages experienced a series of failures and could not be sent before the timeout period.
- ***Connect Time -*** This number indicates the total amount of time (in minutes) used by AMIS networking calls during the time interval indicated.
- ***Failures*** - The number of AMIS messages that were not successfully delivered to other AMIS sites due to specific resource problems. This statistic is further subdivided into the types of problems that may prevent messages from being delivered:
  - ***No Res*** - or *no resources*, means that a modem or voice port could not be accessed to send these messages to another AMIS site.
  - ***Not Reach -*** or *not reachable*, means that the remote AMIS site could not be accessed.
  - ***Prot Error -*** or *protocol error*, means that the connection was made to the remote AMIS site, but message delivery was prevented by a protocol error.

Outcalling Detail

The Outcalling report details outcalling activity for the remote notification and delivery to non-users services (Figure 11-11).

Figure 11-11xxx
The Outcalling Detail report

Operational Measurements

Outcalling Detail (Remote Notification and Delivery to Non-User)

		Number of New Requests		Number of Attempts New Request		Number of Attempts Retries		Number of Successes		Wait Time Avg	Wait Time Max
Interval	Start-End	RN	DNU	RN	DNU	RN	DNU	RN	DNU	(sec)	(sec)
2/08	13:00-14:00	0	0	0	0	0	0	0	0	0	0
2/08	14:00-15:00	1	0	0	0	0	0	1	0	259	259
2/08	15:00-16:00	4	0	1	0	0	0	0	0	0	0
2/08	16:00-17:00	1	1	0	1	0	0	0	0	0	0

Select a softkey >

Exit Current	Exit All		Next Page*	

*Appears when the information fills more than one screen.

The report displays the following fields:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***Number of New Requests*** - The total number of new requests that were made for outcalling services during the interval.
  - ***RN -*** The number of new requests for the remote notification service.
  - ***DNU -*** The number of new requests for the delivery to non-user service.

- ***Number of Attempts*** - The total number of remote notification and delivery to non-user attempts made during the interval.
  - ***New Requests*** - This number represents the number of attempts that have been made to answer the new requests for RN and DNU. If the number of attempts does not equal the Number of New Requests (see the previous field), the system is not keeping up with outcalling requests and more channels may need to be allocated to outcalling.
  - ***Number of Retries*** - This number represents the number of times that the remote notification and delivery to non-users services have retried calls because one of the following occurred at the destination number:
    - the number was busy (RN and DNU)
    - there was no answer (RN and DNU)
    - the call was answered but no messages were retrieved (RN)
    - the required DTMF confirmation was not given (DNU)
- ***Number of Successes*** - The number of successful remote notifications and messages successfully delivered to non-users that have occurred during the interval.

  RN successes are measured in terms of user login. In other words, an RN call is considered successful if the user logs on to his or her mailbox when the notification is received (on the same call as the notification). If the user receives the notification, hangs up and then logs into his or her mailbox, this is not counted as a success since the user terminated the notification call without logging in.

  > ***Note:*** For remote notification to a pager, RN calls are never counted as successful in reports because the paging service cannot log on to the mailbox. A better measure of the effectiveness of RNs to pagers is to compare the number of RN retries to RN attempts. However, bear in mind that an RN retry does not necessarily mean the RN attempt to the paging service failed, it only signifies that the user did not log on within the retry interval.

  A DNU call is considered successful if the called party answers the call (and DTMF confirmation is given if required).
- ***Wait Time*** - These values are an indication of how long the outcalling agent is taking to acquire a channel to outcall to the specified DN.
  - ***Avg (sec)*** - This is the average amount of time, based on all outcalling attempts made during the interval, that it took the outcalling agent to acquire the resources necessary to make the outcall.
  - ***Max (sec)*** - This number represents the outcalling attempt that took the longest amount of time to acquire the resources necessary to make the outcall.

Disk Usage Detail Report

The Disk Usage report provides information on disk space usage on the voice storage volumes (Figure 11-12).

Figure 11-12xxx
The Disk Usage Detail Report

```
                          Operational Measurements
Disk Usage Detail

                     Volume     Voice Volume    Voice Space       Text Space
Interval Start-End    Name      Size (hh:mm)     Used (%)          Used (%)

09/30  10:00-11:00 VS1                  1;51       33                47
                    VS202              33:15       33                17
                    VS203              25:45       10                30
                    VS204              25:45       10                30
09/30  11:00-12:00 VS1                  1:51       33                47
                    VS202              33:15       33                17
                    VS203              25:45       10                30
                    VS204              25:45       10                30
09/30  12:00-13:00 VS1                  1:51       33                47

Select a softkey >

  Exit        Exit                        Next
 Current      All                         Page*
```

*Appears when the information fills more than one screen.

The following fields are displayed:

- ***Interval Start-End*** - Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***Volume Name*** - The name of the user volume (e.g., VS2, VS202, VS203, etc).
- ***Voice Volume Size*** - The amount of disk space that has been used. This is displayed in hours and minutes. One hour of voice storage is equivalent to 8.5 megabytes.

- ***Space Used*** - The percentage of disk space used at the end of the interval.

  Note: The screen shows the Voice Storage and not the Data Storage used. The Data Storage must also be watched. See the section “Volume Administration” in the chapter “General Administration”.

T1 Link Handler Detail

This report gives the number of errors encountered on the T1 links during a given reporting interval.

Figure 11-13xxx
The T1 Link Handler Detail report

```
                         Operational Measurements

T1 Link Handler Detail

                                                   Out of            Backward   Forward
                                          Bipolar  Frame  Extended  Slip       Slip
Interval Start-End      T1 Link          Violatns  Errors SF Errors Count      Count

01/01/92  09:00-10:00  13-1-1              0        0       0        0          0
01/01/92  09:00-10:00  13-1-2              0        0       0        0          0
01/01/92  09:00-10:00  13-1-3              0        0       0        0          0
01/01/92  09:00-10:00  13-1-4              0        0       0        0          0
01/01/92  09:00-10:00  14-1-1              0        0       0        0          0
01/01/92  09:00-10:00  14-1-2              0        0       0        0          0
01/01/92  09:00-10:00  14-1-3              0        0       0        0          0
01/01/92  09:00-10:00  14-1-4              0        0       0        0          0
01/01/92  10:00-11:00  13-1-1              0        0       0        0          0
01/01/92  10:00-11:00  13-1-2              0        0       0        0          0

Select a softkey >

   Exit          Exit                              Next Page*
  Current        All
```

*Appears when the information fills more than one screen.

The following fields are displayed on this screen:

- ***Interval Start-End -*** Data is divided into intervals. The length of the interval depends on the entry made in the *Traffic: Commit Interval* field in the Operational Measurement Options screen. The number of intervals displayed depends on the entries made in the *Traffic Period Start* and *Traffic Period End* fields in the Operational Measurement Options screen. For example, if data is collected 24 hours a day (from 01:00 to 01:00), and the commit interval is one hour, the report will divide the data into 24 intervals for each day included in the report. The amount of data displayed in this report depends on the *Report Start* and *Report End* entries that were made in the Traffic Reports screen. If no report start and end dates and times were given, all data currently stored on disk are displayed.
- ***T1 Link -*** The T1 link for which the reported statistics apply. The link that is shown corresponds to the Link ID in the T1 Link Configuration screen in Hardware Administration.

- ***Bipolar Violatns -*** The number of bipolar violations that have occurred in the specified interval.

  An excessive number of violations indicates one of the following:

  - The quality of the line is poor.
  - The line code between the SPM and the channel bank does not match. Check the line code in the SPM and channel bank.
- ***Out of Frame Errors -*** The number of out of frame errors that have occurred in the specified interval.

  An excessive number of violations indicates one of the following:

  - The quality of the line is poor.
  - The clocking reference between the SPM and channel bank/terminating equipment is poor or is not set up properly.

    If the SPM is configured to provide the timing reference (free-run mode), make sure that the channel banks/terminating equipment derive the timing reference from the SPM.

    If the terminating equipment is configured to provide the timing reference, make sure that you have nominated one or more T1 spans as candidates for clock referencing and that one of the nominated spans is active. For information about nominating T1 spans as candidates for clock referencing see “Modifying the T1 link setup” in the “Hardware Administration” chapter. A candidate is made the active reference in the T1 Link Status screen (described in the “System Status and Maintenance” chapter).
- ***Extended SF Errors -*** Not applicable.
- ***Backward Slip Count -*** The number of backward slips that have occurred in the specified interval. See the description for Out of Frame Errors.
- ***Forward Slip Count -*** The number of forward slips that have occurred in the specified interval. See the description for Out of Frame Errors.

User Usage Reports

The User Usage Report provides statistics for local voice messaging usage on a per-user basis. If Meridian Networking and/or AMIS network usage is installed, then the report also displays users' daily networking activity. Fill in the User Usage Report screen (Figure 11-14) to specify the criteria by which data is to be retrieved in the report.

> ***Note:*** Check the Operational Measurement Options screen to make sure that the collection of user usage data is enabled.

Figure 11-14xxx
The User Usage Reports screen

```
                         Operational Measurements

  User Usage Reports

  Selection Criteria:                [All]  Last_Name  Mailbox  Department

  Sorted:                            Alphabetically By_Department

  Include Local Usage:               No[Yes]

* Include Meridian Network Usage:    [No] Yes

**Include AMIS Network Usage:        [No] Yes


  Report start   (dd/mm/yy): _________________   (or blank for oldest)
  Report end     (dd/mm/yy): _________________   (or blank for newest)

 Select a softkey >

              Cancel        View         Print
                            Reports      Reports
```

```
* Appears when Meridian Networking is installed.
**Appears when AMIS Networking is installed.
```

The following fields are displayed:

- ***Selection Criteria*** - The options that are offered represent search parameters. Any statistics matching your selection will be displayed in the report. Your choices are:
  - ***All Users*** - User usage data for all local users will be displayed in the report.
  - ***Last_Name*** - When selected, you are prompted for the last name of the subscriber whose data you want to view. If the last name is not found, use the Find Users feature in User Administration to verify that the name exists in the system.

- *Department* - When selected, you are prompted for a department name. All users associated with that department will be displayed in the report. The entry you make must correspond to an existing entry in the system. You may use wildcard characters ("+", "?" or "_") to retrieve a group of departments.

  Note: When searching by department, users with blank department fields will not be displayed.

- *Mailbox* - When selected, you are prompted for the mailbox number of the user whose data you want to view. You may use wildcard characters ("+", "?" or "_") to retrieve a range of mailboxes. If the mailbox number is not found, use the Find Users feature in User Administration to verify that the mailbox number exists in the system.

- ***Sorted*** - If your selection criteria is "All Users", you can choose to sort the user data alphabetically, according to user names, or according to department names.

  Note: When sorting by department, users with blank department fields will not be displayed.

- ***Include Local Usage*** - When this field is set to "Y es", the report will include user usage data for local voice messaging. The default is "Yes".
- ***Include Meridian Network Usage*** - When this field is set to "Y es", the report will include user usage data for Meridian networking activity. The default is "No".
- ***Include AMIS Network Usage*** - When this field is set to "Yes", the report will include user usage data for AMIS network activity. The default is "No".
- ***Report Start (dd/mm/yy)*** - The date on which the selected reports are to start. If *Report Start* predates the earliest available date, the report starts with the earliest available date. Leave the field blank to retrieve reports for the earliest available data.
- ***Report End (dd/mm/yy)*** - The date on which the selected reports are to end. If *Report End* exceeds the latest available period, the report ends with the last available period. Leave the field blank to report on the most recent data.

Procedure 11-3xxx
Viewing User Usage Reports

Starting point : The Operational Measurement Reports screen, <3> entered.

The User Usage Reports screen is displayed (Figure 11-14).

1 Choose the selection criteria by which you want to retrieve data.

2 If the selection criteria is “All Users” select how you want the data to be sorted: alphabetically (by user name) or by department name.

3 Select the type of data you want to view: local usage, Meridian networking usage, or AMIS network usage. You can select all three if required.

4 If you wish to specify a start and stop time for the reporting period, enter the required values in the *Report Start* and *Report End* fields.

5 Choose step 5a to view the reports, 5b to print the reports, or 5c to cancel.

a. Use [View Reports].

The selected report screens are displayed (see the following pages for descriptions of each report).

Use [Next Page] to view subsequent pages of the report; use [Exit] to return to the User Usage Reports screen.

b. Use [Print Reports].

You are prompted to make sure your printer is ready and on-line.

Use [Continue Printing] to print the reports, or use [Cancel Printing] at any time to cancel printing (there may be some delay before control is returned to the screen because it waits for the printer to stop printing).

c. Use [Cancel].

The Operational Measurements menu is redisplayed.

Viewing user usage reports

When you view the report on the terminal or from a printout, the data is arranged as shown in Figure 11-15. This figure shows all three types of user usage data (local, AMIS and Meridian networking).

Figure 11-15xxx
The Voice Messaging User Usage report

```
                          User Usage Reports

Last Name    First Name     Department     Mailbox        Billing Class
Smith        David          T20            2255           1

Local Usage:
             Number of      Connect      Number of    Message   Disk
             Sessions       Time         Messages     Length    Used
Date       EM/Ans Logon     (mm:ss)    EM/Ans Logon   (mm:ss)   (mm:ss)
02/12/90      10     4         4:00     9    2          6:30     4:30
02/13/90       8     3         3:12     8    3          12:35    5:30
--------------------------------------------------------------------
Total         18     7         7:12    17    5          19:05

Meridian Networking Usage:
             Number of    Total  Number of    Total  Number of    Total
             Economy      Length Standard     Length Urgent       Length
Date         Messages     (mm:ss) Messages    (mm:ss)Messages     (mm:ss)
02/12/90           12     4:12        10       2:30     6          4:10
02/13/90            8     2:23         7      11:40     0          0:00
                 --------------     --------------    ---------------
Total              20     6:35        17      14:10     6          4:10

AMIS network Usage:
             Number of    Total  Number of    Total  Number of    Total
              Economy     Length Standard     Length Urgent       Length
Date         Messages     (mm:ss) Messages    (mm:ss)Messages     (mm:ss)
02/12/90           10     3:10         1       1:30     0          0:00
02/13/90           10     1:20         7       5:10     0          0:00
                 --------------      -------------    ----------------
Total              20     4:30         8       6:40     0          0:00

----------------------------------------------------------------------
Last Name    First Name     Department     Billing Class     Mailbox
Roeg         Nick           Marketing         1              2929

Local Usage:
             Number of      Connect       Number of      Message   Disk
             Sessions       Time          Messages       Length    Used
Date         EM/Ans Logon   (mm:ss)     EM/Ans Logon     (mm:ss)   (mm:ss)

 Select a softkey >

   Exit                                       Next
                                              Page*
```

*Appears when the information fills more than one screen.

The following fields appear:

- ***Last Name*** - The user's last name.
- ***First Name*** - The user's first name.
- ***Department*** - The user's department name.

- ***Billing Class*** - This field is used for billing purposes. The class number is the number of the model with which the user was added.
- ***Mailbox*** - The user's mailbox number.

The following fields appear for Local Usage:

- ***Date*** - The date of the reporting interval.
- ***Number of Sessions (EM/Ans and Log)*** - The number of express messaging, call answering and logon sessions that occurred during the interval. To check the number of messages that were actually received or created during these sessions, check the *Number of Messages* field.

  If the number of logons is zero, you might want to check the *Time of Last Logon* field in the View/Modify Local Voice User screen. If a considerable amount of time has passed since the last successful logon, you may want to contact the user to see if he or she is having any problems logging on. For example, the user may not know how to log on and retrieve messages (especially if this is a new user) or the user may have forgotten the mailbox password and has stopped trying to log on.
- ***Connect Time*** - The length of time that the user was connected to the voice messaging service on the given date.
- ***Number of Messages*** - The number of messages that the user received and created on the given date.
  - ***EM/Ans*** refers to the number of messages left in the user's mailbox by both the express messaging and call answering services. The number of abandoned calls (where no message is left) can be calculated by subtracting the Number of EM/Ans Messages from the Number of EM/Ans Sessions.
  - ***Logon*** refers to the number of messages that the user created on the report date.
- ***Message Length*** - The total time (in minutes and seconds) of all call answering messages received and messages created by the user on the given date.
- ***Disk Used*** - The amount of storage used by the user (measured in minutes and seconds) on the given date.

The following fields appear for Meridian Networking Usage and AMIS network Usage:

- ***Date*** - The date of the reporting interval.
- ***Number of Economy Messages*** - The number of economy messages that the user created on the given date. This includes both Meridian networking and AMIS networking messages.
- ***Total Length*** - The total length (in minutes and seconds) of all Meridian networking and/or AMIS networking messages created by the user on the given date and tagged as economy.

- ***Number of Standard Messages*** **-** The number of standard messages that the user created on the given date. This includes both Meridian networking and AMIS networking messages.
- ***Total Length*** **-** The total length (in minutes and seconds) of all Meridian networking and/or AMIS networking messages created by the user on the given date and tagged as standard.
- ***Number of Urgent Messages*** **-** The number of urgent messages that the user created on the given date. This includes both Meridian networking and AMIS networking messages.
- ***Total Length*** **-** The total length (in minutes and seconds) of all Meridian networking and/or AMIS networking messages created by the user on the given date and tagged as urgent.

AMIS Networking

Overview

This chapter describes the AMIS (Audio Messaging Interchange Specification) networking protocol and its administration in DMS VoiceMail.

The AMIS protocol is an industry standard which allows users of different vendors' voice messaging products to exchange voice messages. DMS VoiceMail users can send voice messages to users of other voice messaging systems (as long as they support the AMIS protocol), receive messages from other AMIS sites and reply to these messages using standard DMS VoiceMail functionality. The AMIS open access design allows anyone who has access to AMIS to send messages without the need for pre-arranged passwords, site definitions or specialized hardware.

Because the AMIS protocol supports a wide variety of architectures, from the simplest systems to high-end multi-function systems, only the most basic or commonly used features are supported. Therefore, many of the more advanced and sophisticated DMS VoiceMail features cannot be used when communicating AMIS messages.

The following DMS VoiceMail functions are supported by AMIS:

- DMS VoiceMail users can compose voice messages to AMIS recipients. This requires a System Access DN (described later).
- Users can receive messages from other AMIS sites and can use the Reply To feature to respond to these messages immediately.
- Users can forward AMIS messages to other DMS VoiceMail or AMIS users. When a forwarded message is received, the message is preceded with the spoken announcement "attached message". If the message was forwarded several times, this announcement will be played before each attachment.
- Users can tag messages going to AMIS recipients as urgent, standard or economy. Message priorities are discussed in greater detail in the "DMS VoiceMail Network Administration" chapter.
- Acknowledgment tags are supported for AMIS messages but function differently than for non-AMIS messages. For non-AMIS recipients, an

acknowledgement indicates that the message has been listened to whereas for AMIS recipients, it indicates that that the message was delivered to the mailbox.

- Timed delivery is supported.
- AMIS recipients can be mixed with other recipients (local voice users, private network users, distribution lists, non-users) during message composition.
- When messages are not successfully delivered to AMIS recipients, DMS VoiceMail users will receive a non-delivery notification (NDN).
- Retry scenarios for (holding times and stale times) for urgent, standard, and economy messages can be defined by the system administrator.
- Billing records, indicating call length, originator, recipient, and message length will be generated after each AMIS message session. A set of Operational Measurement reports will be provided, similar to those used for Meridian networking. These reports can be downloaded for further processing through AdminPlus.

The following DMS VoiceMail feature is not supported by the AMIS protocol:

- Private Message tags have no effect on AMIS messages. Messages tagged as private are not sent to any AMIS address in the message envelope, but will be returned to the originator with an NDN. (The message will be delivered as a private message to all other recipients in the envelope). This is done because there is no way to prevent private messages from being forwarded and therefore violating the originator's intent. Users familiar with Meridian networking should be informed that they cannot tag AMIS messages as private.

The features listed below are not typically supported by the AMIS protocol. However, they can be made available for a customer group if Meridian Networking is enabled. This is achieved by adding AMIS sites as remote sites in the Meridian network, thus creating a *virtual node* for each remote AMIS site. This is described in the section "Configuring AMIS sites as virtual nodes in a Meridian private network" on page 12-12. When AMIS sites are configured as virtual nodes (and when the users at those sites are added as remote voice users), the following features become available:

- Personal and System Distribution Lists
- Name Addressing
- Personal Verification and Call Sender for call answering messages
- Personal Verification, Call Sender and Reply To for voice messages

  Note: Some of the above features have additional requirements which are detailed on page 12-12.

Users familiar with Meridian networking should be informed of features (used to compose messages) that are not supported by the AMIS protocol. This is especially recommended for features such as message privacy where messages will not be delivered.

Configuring the AMIS service

As administrator, you are responsible for the configuration and specification of the operational characteristics of the AMIS Networking service.

The following sections detail the steps necessary to configure the AMIS service.

Identify which service will accept AMIS calls

Incoming AMIS networking calls must terminate on one of the following types of service DNs:

- a special DN defined for AMIS in the VSDN table,
- a voice menu DN that is defined in the VSDN table, or
- a thru-dialer DN that is defined in the VSDN table.

A dedicated line DN can be created for the AMIS service on the DMS/SL-100 although this is not necessary. This is because both voice menus and thru-dialers can accept inbound AMIS Networking calls. The only requirement is that the voice menu or thru-dialer be provisioned with DID access (i.e., must be directly dialable). Otherwise, you will have to create a line DN specifically for the AMIS service. When an inbound AMIS call terminates on a voice menu or thru-dialer, it is recognized as an AMIS call and an AMIS Networking session is initiated. (Note that for this to work, the field *Act on AMIS Initiation Tone* in the Voice Services Profile screen must be set to “Yes”. This is described in the following sections.)

Using a voice menu to accept inbound AMIS calls

If you are going to use a voice menu to accept AMIS calls, carry out the following steps. (If the voice menu application already exists, begin at step 3.)

On the DMS/SL-100

1 If there are no available line DNs on the switch, create one for the voice menu application you are about to create. See the section "Configuring voice services" in the "Voice Administration" chapter for details.

In DMS VoiceMail

2 Build the voice menu application. Voice menus are described in the "Voice Administration" chapter.

3 In the Voice Menu Definition, set the Initial No Response action as RP (for Repeat Menu Choices). This is necessary to ensure that a call will remain connected to the voice menu for at least 10 seconds, otherwise the call may be prematurely disconnected. It takes about 10 seconds for the voice menu to get a signal from AMIS and then transfer the call to the AMIS service. By the time the menu choices are repeated a second time, 10 seconds will have passed and the call will have been transferred. Voice Menus are described in the "Voice Administration" chapter.

4 In the Voice Services Profileset the field *Act on AMIS Initiator Tone* to "Yes", otherwise AMIS calls that are placed to the voice menu will not be transferred to the AMIS service.

5 Enter the voice menu DN in the VSDN table.

> ***Note:*** It is recommended that you use a voice menu to accept AMIS calls rather than a thru-dialer. This is because the *Short Disconnect* field also affects voice menus. You cannot configure a separate value for voice menus and thru-dialers. Therefore, if you use a thru-dialer to accept AMIS calls, you will not be able to set this value lower than 10 seconds (which you may decide is too long for voice menus).

Using a thru-dialer to accept inbound AMIS calls

If you are going to use a thru-dialer to accept AMIS calls, carry out the following steps. (If the thru-dialer already exists, begin at step 3.)

1 If there are no available line DNs on the switch, create one for the thru-dialer you are about to create. See the section “Configuring voice services” in the “Voice Administration” chapter for details.

2 Build the thru-dialer application. Thru-dialers are described in the “Voice Administration” chapter.

3 In the Voice Services Profileset the *Short Disconnect* field to a value of at least 10 seconds. This field determines how long the system will wait for an initial response (keypad entry) before disconnecting the call. Since it takes 10 seconds for an AMIS call to be transferred from a thru-dialer to the AMIS service, AMIS calls will be prematurely disconnected if this field is set to a value less than 10.

4 In the Voice Services Profileset the field *Act on AMIS Initiator Tone* to “Yes”, otherwise AMIS calls that are placed to the thru-dialer will not be transferred to the AMIS service.

5 Enter the thru-dialer DN in the VSDN table.

Creating a special AMIS service DN

If you will not be using a voice menu or thru-dialer to accept AMIS calls, you will have to create a special DN for the AMIS service.

On the DMS/SL-100

1 If there are no available line DNs on the switch, create one for the AMIS service. See the section “Configuring voice services” in the “Voice Administration” chapter for details.

In DMS VoiceMail

2 Add the DN for the AMIS service to the VSDN table.

Configure AMIS networking information

You are now ready to configure the parameters specific to the AMIS networking service. The following parameters are configured in the View/Modify AMIS Networking Information screen (Figure 12-3).

In this screen you will have to specify the following.

AMIS compose prefix

This is the number that is used by users at the local site to send AMIS messages to remote sites. It is entered during message composition to indicate that the address the user is entering is an AMIS address. You will have to inform the users at the local site of this prefix.

System access number

This DN identifies the local site within an AMIS network. This is the DN to which messages will be addressed by users at remote AMIS sites. Publish this number as your site's AMIS number. The system access number includes the following elements:

- the country code of the local site, up to 4 digits in length;
- the area code of the local site, up to 4 digits in length;
- the local number of the local site.

  The local number must terminate on the DN that has been defined in the VSDN table - the DN of the voice menu, thru-dialer or AMIS service that will be used to accept incoming AMIS calls.

Message priorities and thresholds

A user can assign one of three priorities to an AMIS message: Economy, Standard, and Urgent. Economy priority messages are sent at a specified time each day. This is referred to as the *initiation time*. For Standard and Urgent messages, you can specify a *holding time* - the length of time that messages are retained before they are sent to remote sites. Urgent messages are assigned shorter holding times, and are therefore sent more often than Standard messages.

The timing of message delivery is determined by a series of thresholds that are assigned specific values. The following sections describe the operation of thresholds.

Holding Time Threshold

The AMIS Networking service does not set up a delivery connection every time a message destined for a remote site is sent by a local user. Instead, to reduce costs, each message is placed in a queue to await the submission of more messages for delivery. This threshold applies only to Urgent and Standard priority messages, not Economy messages.

When the system *wakes up* (see the description of “Wakeup Interval”) it checks for AMIS messages waiting to be sent. If there are any AMIS messages, the system then checks the batch threshold. If this threshold has been reached (i.e., if this number is set to 10 and there are 11 messages they will immediately be sent). If the batch threshold has not been reached the messages are placed in a send queue. When either the standard or urgent holding time has been reached, all standard and urgent messages are sent.

For example, a user submits a standard message at 1:00 p.m. The standard holding time is 1 hour. The message is retained until 2:00 p.m. awaiting further messages destined to the same site. At 1:15 p.m. a user sends an urgent message and the urgent holding time is 15 minutes. At 1:30 p.m. the urgent message is eligible for delivery. The next time the system wakes up, it will place both messages in a send queue (if the batch threshold has not been reached). At 1:30, the urgent holding time, a network connection is established to each site to which a message is destined. Since a connection now exists, the standard message is transferred along with the urgent message.

Economy messages, on the other hand, are always delivered at a specific time (for example, 6:00 p.m. every evening) and are therefore unaffected by the holding time threshold. Economy messages will not be delivered until the absolute time, regardless of whether or not other urgent or standard messages are ready to be delivered. This preserves the overnight delivery nature of economy messages.

Stale Time Threshold

To prevent the AMIS Networking service from retaining messages that cannot be delivered because of local or remote site problems, a *stale time* is defined for each message priority. If a message is still undelivered after the specified stale time interval, the sender of the message receives a non-delivery notification (NDN) indicating that the message has not been transferred within the time limit specified for its priority. This is known as *stale dating* and prevents the AMIS Networking service from becoming congested with undeliverable messages (if, for example, the site has been disabled for maintenance). Messages that are undelivered must be recomposed and the user must send them again.

Wakeup Interval

The AMIS Networking service wakes up at periodic intervals to check if there are messages to be sent (the standard and urgent holding times and the batch threshold are checked). You can set this interval according to the system’s specific needs. For lightly loaded systems with many remote sites requiring long distance calls, the intervals should be longer, for example, 15 to 30 minutes. For heavy traffic systems, such as those needing only local calls to reach remote sites, the interval may be shorter, for example, 2 to 10 minutes.

Restrictions

There are several types of restrictions that you can place on the operation of AMIS Networking.

Time restrictions

You can create two time windows, one for weekdays and one for weekends, that define the hours during which AMIS messages are allowed to be delivered. You will have to check with the regional legislation regarding computer-generated phone calls to establish when you are prohibited from sending electronic messages.

Temporary feature disable

You can temporarily restrict users from accessing the AMIS Networking service. This may be necessary to prevent system abuse or to clear the system of messages that cannot be delivered and are tying up resources. Check the Networking Status screen to see if a large number of AMIS messages remain queued for an extended period of time. This indicates that DMS VoiceMail is unable to send messages due to a local or remote problem. If this is the case, you may have to disable AMIS until the problem has been resolved.

Enabling AMIS for your users

If you are adding new users

Before you add users to the system, it is recommended that you create a new user model in which AMIS networking is enabled. The default user models do not give users the ability to receive or compose/send AMIS messages.

Restriction and permission codes are applied to the AMIS messages that local users send. If you want users to be able to send AMIS messages to sites that are long distance, verify that the long distance dialing prefix (“91” for example) is not defined as a restriction code. Remember also that these codes are intended to prevent abuse of the system. The restriction codes should specify the numbers to which users are not qualified to send AMIS messages.

By default, the “Local” set of restriction/permission codes are applied to AMIS networking messages. Ensure that you know which restriction/permission code set you want to apply to outgoing AMIS messages and then make sure that set is selected in the user model.

To create a user model that enables AMIS networking for new users:

1 Check the restriction/permission codes that are defined in the Voice Security Options screen. Determine which set of codes you want to apply to outbound AMIS messages. If necessary, modify one of the existing sets.

See the section “Voice Security Options” in the “Voice Administration” chapter for more information.

2 Select View/Modify User Models from the User Administration Menu.

3 In the View/Modify User Models screen:

a. Set *Receive AMIS Messages* to “Yes”.

b. Set *Compose/Send AMIS Messages* to “Yes”.

c. Select the AMIS restriction/permission codes that are to apply to outbound AMIS networking messages.

For more information, see the section “User models” in the “User Administration” chapter.

Enabling AMIS for existing users

If you have upgraded to Release 8 from a previous release, you will have to enable AMIS for each existing user. You must also check that the restriction/permission codes that have been applied to outbound AMIS messages are appropriate. This can be done from the View/Modify Local Voice User screen as described in the “User Administration” chapter.

To enable AMIS networking for an existing user:

1 Check the restriction/permission codes that are defined in the Voice Security Options screen. Determine which set of codes you want to apply to outbound AMIS messages. If necessary, modify one of the existing sets. It is important to apply the correct codes in order to prevent abuse of the system. The restriction codes specify the numbers to which the user is not qualified to send AMIS messages. By default the code set named “Local” is applied. If you want the user to be able to send AMIS messages to long distance addresses, ensure that the long distance dialing prefix (“91” for example) is not restricted.

See the section “Voice Security Options” in the “Voice Administration” chapter for more information.

2 Access the user profile. From the Main Menu, Select User Administration, followed by View/Modify Local Voice User.

For each user:

3 Set *Receive AMIS Messages* to “Yes”.

4 Set *Compose/Send AMIS Messages* to “Yes”.

5 Select the AMIS restriction/permission codes that are to apply to outbound AMIS networking messages.

Networking call maximum

Determine the maximum number of outgoing networking calls that can be made simultaneously. If too many calls are allowed, you may severely limit the resources that are available for other DMS VoiceMail services.

Dialing prefix for long distance calls within your area

This is the prefix that is required to make long distance calls to DNs sharing the same area code as the local DN. For example, “1” or “1 416”.

Configuring AMIS sites as virtual nodes in a Meridian private network

If Meridian Networking is installed, a remote AMIS site can be added to your Meridian network as a *virtual node*. This makes AMIS Networking transparent to local users when they address messages to users at remote AMIS sites. Normally, when addressing a message to an AMIS site, a local user enters an AMIS prefix followed by a full dialing code which can include a country code and area code and always includes a local number and mailbox number (see the next section). By configuring an AMIS site as a virtual node, local users enter the address in the same format they use to address a message to a user at a remote Meridian networking site.

The following features are available to local users only if the remote AMIS site they are addressing the message to is configured as a virtual node. Most of these features also require that the remote users be added to your system as remote voice users (through User Administration).

For more information about configuring an AMIS site as a virtual node, see the chapter “Meridian Networking Administration”.

Personal distribution lists and system distribution lists

The following requirements must be met if you you want local users to be able to add AMIS recipients to personal distribution lists and if you (the administrator) want to be able to add them to system distribution lists.

- The AMIS site is configured as a virtual node.
- The remote users at that site have been added as remote voice users through User Administration.

Name dialing and name addressing

Local users can use name dialing and name addressing when calling/sending messages to remote AMIS users if the following requirements are met:

- The AMIS site is configured as a virtual node.
- The remote users at that site have been added as remote voice users through User Administration.

Personal verification and call sender for voice messages

Local voice users will:

- hear a personal verification when a remote AMIS user leaves voice message in the local user’s mailbox, and
- be able to use call sender to reply to the voice message,

if the following requirements are met:

- The AMIS site is configured as a virtual node.
- A call sender prefix has been configured (for the call sender feature).
- The remote users at that site have been added as remote voice users through User Administration.

- The remote voice users' mailbox numbers must be the same as their extensions at the remote site. (This requirement is for call sender only.)

Personal verification, call sender and reply-to-sender for call answering messages

Local voice users will:

- hear a personal verification when a remote AMIS user leaves a message during a call answering session,
- be able to use reply to sender in response to the message, and
- be able to use call sender,

if the following requirements are met:

- The AMIS site is configured as a virtual node.
- A call sender prefix has been configured (for the call sender feature).
- The remote users at that site have been added as remote voice users through User Administration.
- The Calling Line ID (CLID) must be present.

Addressing AMIS messages (to non-virtual nodes)

When a user composes a message that is destined for an AMIS site that is not defined as a virtual node, he or she begins by entering the AMIS Compose Prefix. This informs the system that the address that is about to be entered is that of an AMIS site.

The prefix is followed by the access code that is required to dial out of the DMS VoiceMail system. This will either be an international access code (if the recipient is in another country), a long distance access code such as “91” (if the recipient is in the same country but a different area code), or a local access code such as “9” (if the recipient is in the same country and area code).

The user then enters the System Access DN. This DN identifies the AMIS site to which the message will be delivered. This number includes the following elements:

- the country code of the remote site, up to 4 digits in length (optional if identical to the country code of the sending system);
- the area code of the remote site, up to 4 digits in length (optional if it is identical to the area code of the sending system);
- the local number of the remote site

After entering the local number, the user enters “#” (number sign). In summary, the number needed to address an AMIS site is entered in the following format:

<AMIS compose prefix><access code><System Access DN>#

After pressing “#” the user is prompted to enter the mailbox number. The following recording is played: *“Enter the mailbox number for this Open Network user followed by number sign”*.

> ***Note:*** Before an AMIS message is played to a recipient, the sending system plays the following prompt: *“Open Access computer message, press 1 to cancel”*. This enables someone who has answered a call to a wrong number to disable further calls. If a recipient cancels message delivery in this manner, all messages currently queued to that number will be returned to their respective senders and further deliveries to the same access DN will be prevented.

The Network Administration Menu

Network Administration allows you to perform administrative and maintenance tasks for the AMIS networking service. The Network Administration menu is displayed when the Network Administration item is selected from the Main Menu.

> **CAUTION**
> **Overnight system audits**
> You should not leave the administrative terminal in any Network Administration menu overnight or important system audits may fail due to a lack of available memory.

Figure 12-1xxx
The Network Administration menu for systems with AMIS Networking only

```
                         Network Administration


     1 View/Modify AMIS Networking




Select an item >
   Exit
```

If both AMIS networking and DMS VoiceMail networking are installed on the system, the following menu is displayed.

Figure 12-2xxx
The Network Administration Menu (for systems with both DMS VoiceMail Networking and AMIS Networking)

```
                    Network Administration

      1 Meridian Mail Network Administration

      2 AMIS Network Administration

_______________________________________________________________

 Select an item >

   Exit
```

Note: DMS VoiceMail networking administration is discussed in Chapter 13. It is only available on CPE systems.

Procedure 12-1xxx
Using the Network Administration menu

Starting point: The Main Menu

1 Select Network Administration.

The Network Administration menu is displayed.

2 Select View/Modify AMIS Networking or AMIS Network Administration.

3 Select [Exit] when you are ready to exit Network Administration.

The Main Menu is displayed.

Modifying AMIS Networking Information

Parameters that control the AMIS networking service are configured in the View/Modify AMIS Networking Information screen (see Figure 12-3).

Figure 12-3xxx
The View/Modify AMIS Networking Information screen

```
                         Network Administration

View/Modify AMIS Networking Information

AMIS Compose Prefix:

Outgoing Messages      Disabled     [Enabled]
Incoming Messages      Disabled     [Enabled]

System Access Number
Country Code           ________
Area/City Code         ___________________
Local Number

Outgoing Messages allowed on weekdays from (hh:mm) ___ to (hh:mm) __
Outgoing Messages allowed on weekends from (hh:mm) ___ to (hh:mm) __
Wakeup Interval (minutes)  ________
Batch Threshold            ________
Networking Call Maximum                 ________
Economy Class Initiation Time (hh:mm)   ________
Economy Class Stale Time (hh:mm)        ________
Standard Class Holding Time (hh:mm)     ________
Standard Class Stale Time (hh:mm)       ________
Urgent Class Holding Time (hh:mm)       ________
Urgent Class Stale Time (hh:mm)         ________

Prefixes for dialing out of this Site
Public Dialing           ____________________________
Long Distance Dialing    ____________________________
International Dialing    ____________________________

Co-resident Office Codes required:    No    Yes
Dialing Prefix for Long Distance Calls within your Area: ______________

 Select an item >

   Save        Cancel                               Co-resident
                                                     Prefixes*
```

*   This softkey is displayed only if the Co-resident Office Codes required field is set to “Yes”.

The following fields are displayed:

- ***AMIS Compose Prefix*** **-** This is the number that users dial to gain access to the AMIS networking service. Make sure that this prefix does not conflict with other network data such as ESN or CDP dialing codes. (There is a conflict if the first two digits of a DN match this prefix.)

- ***Outgoing Messages -*** This field allows you to temporarily prohibit DMS VoiceMail users from sending AMIS messages. The default is “Enabled”. Users who originate messages while transmission is prohibited will immediately receive a non-delivery notification.
- ***Incoming Messages -*** This field allows you to temporarily prohibit incoming AMIS messages from being delivered at this site. The default is “Enabled”.
- ***System Access Number -*** This number identifies your system to other AMIS sites. It is sent along with messages originated at your site and is used when a message is replied to (with an equivalent of the Reply feature) by the recipient. This is also the number that will be used when AMIS messages are addressed to your site. The number consists of the following elements:
  - ***Country Code -*** This is the local site’s country code. This number will be a maximum of 4 digits in length.
  - ***Area/City Code -*** This is the local site’s area code. This number will be a maximum of 4 digits in length.
  - ***Local Number -*** This is the number that is published to the public as the AMIS number. This number must terminate on the extension defined for AMIS in the VSDN table.
- ***Outgoing Messages allowed on weekdays -*** Users are allowed to send messages during the time specified here. Enter the start and end time of the allowed weekday period in the format hh:mm. This may be necessary to comply with regional legislation regarding delivery of electronic messages. You may enter a value from 00:00 to 23:59. The default is 00:00. Users should be notified of restricted hours.
- ***Outgoing Messages allowed on weekends -*** Users are allowed to send messages during the time specified here. Enter the start and end time of the allowed weekend period in the format hh:mm. You may enter a value from 00:00 to 23:59. The default is 00:00. This may be necessary to comply with regional legislation regarding delivery of electronic messages. Users should be notified of restricted hours.
- ***Wakeup Interval -*** The value entered in this field determines how often the system checks for queued messages and sets up the connections required to send those messages. Enter a value in the format mm. The default is 3 minutes. You may enter a value in the range 1 to 99.
- ***Batch Threshold -*** The value entered in this field specifies the total number of standard and urgent messages that can accumulate before delivery commences. The maximum is 99 and the default is “20”.

- ***Networking Call Maximum -*** The value entered here specifies the maximum number of simultaneous outgoing networking calls permitted. If this maximum is reached, no new outgoing sessions will be attempted. This prevents AMIS from using too many resources and interfering with the effective functioning of other DMS VoiceMail services. The default is “4”. The maximum allowable value is “999”.
- ***Economy Class Initiation Time -*** This field determines the time at which AMIS messages tagged as economy are delivered. Enter the time in hours and minutes in the range 00:00 to 23:59.
- ***Economy Class Stale Time -*** The value entered in this field determines the maximum retention time for AMIS messages tagged as economy. When this threshold is reached, a non-delivery notice is sent to the originator and the message has to be composed and sent again. Enter the time in hours and minutes in the range 03:00 to 99:59. The default is “06:00”.
- ***Standard Class Holding Time -*** The value entered in this field specifies the minimum retention time for AMIS messages tagged as standard. This determines the length of time that a standard message is retained before the system attempts to send it. A message may be transferred before this holding time expires if a connection is established for another reason, such as delivering Urgent messages. Enter the time in hours and minutes in the range 00:00 to 33:20. The default is “03:00”.
- ***Standard Class Stale Time -*** The value entered in this field specifies the maximum retention time for AMIS messages tagged as standard. If a message is not delivered before this time, a non-delivery notice is sent to the originator. These messages have to be composed and sent again. The time is entered in hours and minutes and must be in the range 00:00 to 99:59. This value must be at least three times the standard class holding time. The default is “09:00”.
- ***Urgent Class Holding Time -*** The value entered in this field determines the minimum retention time for AMIS messages tagged as urgent. This determines the length of time that an urgent message is retained before the system attempts to send it. The time is entered in hours and minutes and must be in the range 00:00 to 33:20. The default is “00:30”.
- ***Urgent Class Stale Time -*** The value entered in this field is the maximum retention time for AMIS messages tagged as urgent. If a message is not delivered before this time, a non-delivery notice is sent to the originator. These messages have to be composed and sent again. The time is entered in hours and minutes and must be at least three times the urgent class holding time. The valid range for this field is from 00:00 to 99:59. The default is “01:30”.
- ***Prefixes for dialing out of this Site -*** In the following fields, enter the dialing codes that are required to place external calls.

- ***Public Dialing -*** This is the number required to make local external calls. For example, a commonly used dialing code is “9”.
- ***Long Distance Dialing -*** This is the number required to make long distance calls. For example, “91”.
- ***International Dialing -*** This is the number required to make international calls. For example, “9011”.

- ***Co-resident Office Codes required -*** This field should be enabled if there are office code prefixes in your region that share the same area code as your DN yet require the long distance access code for dialing. For example, 766xxxx, 598xxxx, and 602xxxx may all be in the area code 416, but must be dialed using the 1+xxxxxxx format. In larger metropolitan areas there may a large number of these codes possible. The administrator therefore has the option of entering the office codes that should *not* be dialed using the long distance format.

  When this field is set to “Yes” the [Co-resident Prefixes] softkey is displayed. When this softkey is pressed, the Co-Resident Prefixes screen in which these codes are entered is displayed (see Figure 12-4 on the following page). The default is “No”.

- ***Dialing Prefix for Long Distance Calls within your Area -*** This is the prefix that is required to make long distance calls to DNs sharing the same area code as the local DN. For example, “1” or “1 416”.

Procedure 12-2xxx
Configuring AMIS networking information

Starting point: The Main Menu

1 Select Network Administration.

The Network Administration menu is displayed.

2 Select View/Modify AMIS Networking.

The View/Modify AMIS Networking Information screen is displayed (Figure 12-3).

3 Modify the necessary fields.

4 Press [Co-Resident Prefixes] if *Co-Resident office codes* is set to “Yes”.

The Co-Resident prefixes screen is displayed (Figure 12-4). See the following section for more information.

5 To save the configuration, go to step 5a. To exit the screen without saving your changes, go to step 5b.

a. Press [Save].

The data entered in the screen, provided all mandatory fields have been filled in, is saved. The Network Administration menu is displayed.

b. Press [Cancel].

Any changes that you have made are not saved and the Network Administration menu is displayed.

Co-resident prefixes

In the Co-Resident Prefixes screen (Figure 12-4), enter the office codes in your region that share your area code yet are considered long-distance. These codes can be up to 4 digits in length. When you fill up a row with office codes, press [More Fields] to get another row of fields.

Figure 12-4xxx
The Co-Resident Prefixes screen

```
                         Network Administration

 Co-Resident Prefixes

 Enter Co-Resident Office Codes within the same Area/City

 766_   602_   ____   ____   ____   ____   ____   ____


 Save      Cancel                                More
                                                 Fields
```

Procedure 12-3xxx
Configuring AMIS networking information

Starting point: The Main Menu

1 Select Network Administration.

The Network Administration menu is displayed.

2 Select View/Modify AMIS Networking.

The View/Modify AMIS Networking Information screen is displayed (Figure 12-3).

3 Press [Co-Resident Prefixes].

The Co-Resident prefixes screen is displayed (Figure 12-4).

4 Enter the necessary co-resident prefixes.

If you require more fields to enter additional codes, press the [More Fields] softkey.

5 To save the prefixes, go to step 5a. To exit the screen without saving your changes, go to step 5b.

a. Press [Save].

The prefixes are saved. The View/Modify AMIS Networking Information screen is displayed.

b. Press [Cancel].

Any changes that you have made are not saved and the View/Modify AMIS Networking Information screen is displayed.

Meridian Networking

Overview

Note: Meridian Networking is only available on CPE systems.

This chapter provides an overview of Meridian Networking concepts and describes the procedures required to administer the Networking service on your DMS VoiceMail system.

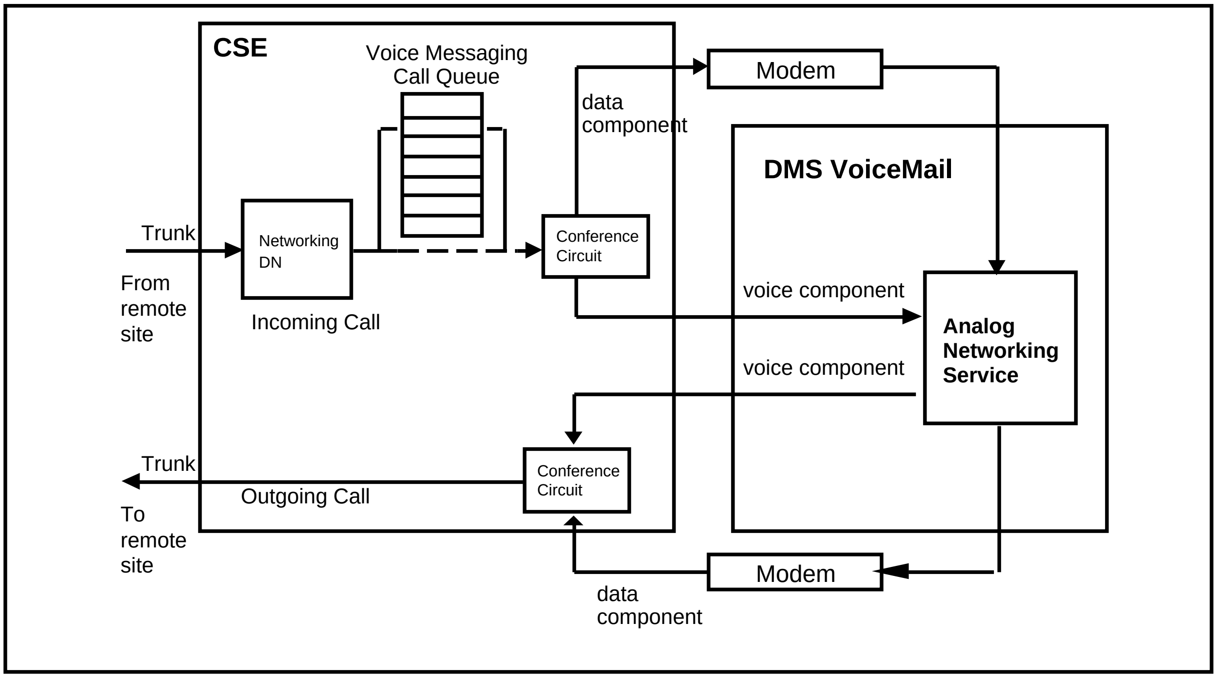
Meridian Networking is an incremental service, installed after the DMS VoiceMail Voice Messaging service is functioning. It is a proprietary networking system developed by Northern Telecom which allows users at your DMS VoiceMail site you to send and receive voice messages, reply to voice messages, and forward voice messages to users located at other DMS VoiceMail sites.

Networking sites

A Meridian network can have up to 50 remote sites. Sites in the network are connected to each other through the long distance network, Direct Distance Dialing (DDD), tie lines, or digital trunks. Message delivery to remote sites in the network is determined by three things: the user-specified priority of the given message (urgent, standard, or economy), the information you specify in the Voice Messaging Options screen (see “Voice Messaging Options” in the “Voice Administration” chapter), and the routing information you specify in the Network Administration screens described in this chapter.

DMS VoiceMail uses the Networking voice service to transfer voice messages to remote sites. The Networking service establishes the call, and connects to the Networking service on the remote system. Control information such as connection passwords, message headers, and message delivery acknowledgements are transferred as data by conferencing modems onto the voice path at the local and remote sites. See Figure 13-1.

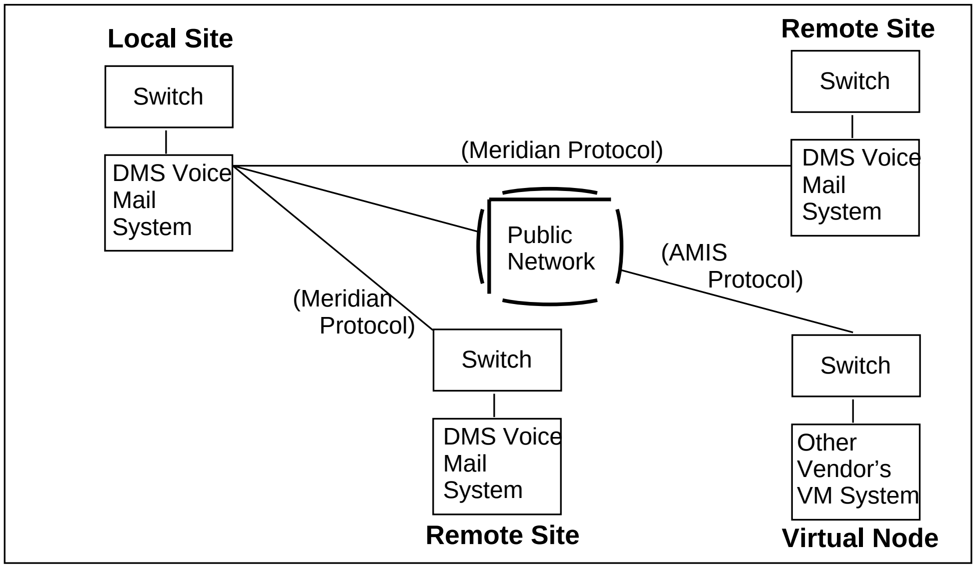
Figure 13-1
Typical networking connection



Adding AMIS sites as virtual nodes

If AMIS networking is installed at the local site, you can add other AMIS sites that use voice messaging systems other than DMS VoiceMail to your network database. AMIS sites that are added as remote sites to your network are referred to as *virtual nodes*.

Figure 13-2xxx
Network with remote DMS VoiceMail sites and virtual nodes



Virtual nodes are transparent to local voice users. This means that local users can address messages to users at virtual nodes as if they were addressing a message to a user at a remote DMS VoiceMail site. Addressing messages to AMIS sites (that are not virtual nodes) is more cumbersome since an AMIS prefix has to entered, followed by any necessary country and/or area codes. These codes are then followed by a local number (typically 7 digits) and finally the mailbox number is entered.

The following features become available if:

1 You configure the AMIS site as a virtual node.

2 You add the users at the AMIS site as remote voice users (through User Administration).

- Local users can include AMIS recipients in personal distribution lists.
- AMIS recipients can be included in system distribution lists.
- Users at the local site can use name dialing to call users at the remote site and name addressing when composing messages.

- Messages from AMIS users that are left during call answering sessions include the personal verification. The local user can use call sender or reply-to-sender to respond. (The CLID must be present for these features to work.)
- When AMIS users send voice messages to local users, the personal verification is included and the local user is able to use call sender to respond to the message. In this case, call sender has an additional requirement. The mailbox numbers at the remote AMIS site must be the same as the extensions at that site.

To configure an AMIS site as a virtual node, add it as a remote site to your network database. This is described in the section “Remote Site Maintenance” beginning on page 13-22.

Message priorities

Messages can be assigned one of three priorities: Economy, Standard, and Urgent. Economy priority messages are sent at a specified time each day. This is referred to as the *initiation time*. For Standard and Urgent messages, the administrator can specify a *holding time* - the length of time that messages are retained before they are sent to remote sites. Urgent messages are assigned shorter holding times, and are therefore sent more often than Standard messages. The initiation time and holding times are defined in the Network Scheduling Parameters screen (see page 13-35).

Thresholds

The timing of message delivery is determined by a series of thresholds that are assigned specific values in the Network Administration screens. The following sections describe the operation of thresholds.

Holding Time Threshold

Networking does not set up a delivery connection every time a message destined for a remote site is sent by a local user. Instead, to reduce costs, each message is retained for a period while awaiting the submission of more messages for delivery to the same remote site. This threshold applies only to Urgent and Standard priority messages, not Economy messages.

Even when the holding time threshold is reached, messages are not immediately sent to the remote site. Instead, they are placed in a queue until the system *wakes up* (see the description of “Wakeup Interval” on the next page). It is at this time that a connection to the remote site is actually established and messages that are eligible for delivery are sent.

A message may be transferred before this holding time expires if a connection is established for another reason. For example, a user submits a Standard message at 1:00 p.m. The Standard holding time is 1 hour. The message is retained until 2:00 p.m. awaiting further messages destined to the same site. At 1:15 p.m. a user sends an Urgent message and the Urgent holding time is 15 minutes. At 1:30 p.m. the Urgent message is eligible for delivery. The next time the system wakes up, a network connection is established to each site to which the message is destined. Since a connection now exists, the Standard message is transferred along with the Urgent message.

Messages sent from the local site can also be transferred on a connection that is initiated from a remote site. This allows messages to be transferred both ways on the same call. This activity is known as *message piggybacking*.

Economy messages are always delivered at a specific time (for example, 6:00 p.m. every evening) and are therefore unaffected by the holding time threshold. Economy messages will not be delivered until the absolute time, regardless of whether or not other Urgent or Standard messages are ready to be delivered or a piggyback connection exists. This preserves the overnight delivery nature of Economy messages.

Stale Time Threshold

To prevent Networking from retaining messages that cannot be delivered because of local or remote site problems, a *stale time* is defined for each message priority. If a message is still undelivered after the specified stale time interval, the sender of the message receives a non-delivery notification (NDN) indicating that the message has not been transferred within the time limit specified for its priority. This is known as *stale dating* and prevents Networking from becoming congested with undeliverable messages (if, for example, the site has been disabled for maintenance). Messages that are undelivered must be recomposed and the user must send them again.

Wakeup Interval

The Networking service wakes up at periodic intervals to check if there are messages to be sent. The interval can be set by the administrator according to the system's specific needs. For lightly loaded systems with many remote sites requiring long distance calls, the intervals should be long, for example, 15 to 30 minutes. For heavy traffic systems, such as those needing only local calls to reach remote sites, the interval may be short, for example, 2 to 10 minutes.

Batch Threshold

This threshold is designed to handle burst conditions that may arise during busy hours. If a large number of Standard and/or Urgent messages are submitted to the Networking service in a period shorter than the holding time, delivery connections are established to off-load the influx. The batch threshold specifies the maximum number of Urgent and Standard messages that can be in the queue to any given site before a connection to that site is attempted. Economy messages are not subject to this threshold.

Maximum Number of Calls

The number of outgoing networking calls can be limited by the administrator. This may be desirable when, for example, there is a busy system with three modems. The administrator can limit the outgoing networking calls to two modems, leaving the third one available for incoming calls.

Administrator responsibility

The administrator is responsible for the configuration and specification of the operational characteristics of the Meridian Networking service.

The administrator is also responsible for the local site's view of the network. Responsibilities include local site administration, through which the basic networking parameters of the local site are specified, and remote site administration, through which the connection parameters from the local site to selected remote sites are specified.

In some installations, one administrator may be designated to maintain the networking parameters of more than one site using dial-up access to these sites. The details of remote access are described in the "Administrator logon and the main menu" chapter.

Each site in a DMS VoiceMail network must be assigned a unique site ID within the range of 1 to 500. Site administrators should coordinate the assignment of these site IDs to ensure that the IDs do not conflict with mailbox numbers within their network. These site IDs are known only to system administrators and their delegates and are used when defining local and remote sites.

Configuring the Networking service

Once the hardware necessary for Networking has been installed on both the CSE and the DMS VoiceMail system, configure the Networking voice service, and then use the Network Administration screens to specify the operational characteristics of the Networking service on your system. See "Voice System Configuration" in the "Voice Administration" chapter for information about creating UCD queues, filling in the Voice Service-DN Table and the Channel Allocation Table.

Networking Directory Number (DN)

In an SL-100 configuration, the Networking service is assigned a directory number that is forwarded to the Voice Messaging service, or the Networking service may be given a dedicated UCD queue. See "Voice System Configuration" in the "Voice Administration" chapter.

Voice Service-DN Table

After the UCD queue has been set up, you must enter the Networking service DN in the Voice Service-DN Table. See "Voice System Configuration" in the "Voice Administration" chapter.

Channel Allocation Table

The Channel Allocation Table (CAT) requires updating only if certain channels are to be dedicated to Networking. To dedicate channels, create a UCD agent queue containing the dedicated voice channels, and enter **NW** as the service abbreviation for these channels in the Channel Allocation Table.

See the section "The Channel Allocation Table" in the "Voice Administration" chapter.

Directory entries

Networking does not require remote mailbox users to be defined in the local directory before local users can address messages to them. Frequently addressed remote users may be added as remote voice users to make it more convenient to address messages to them; see "Adding remote voice users" in the chapter "User Administration". If this is done, the user's DN and spoken name may also be included. The user's name should be recorded with an associated directory number. For example, *"Dexter Smith at 6-555-1213"*. Recording the remote user's spoken name provides a personal verification that the remote mailbox number is correct when addressing or playing back a message. The remote user's directory entry is used to support the Call Sender feature in Voice Messaging. See "Directory Entry Users" in the "User Administration" chapter for details.

Networking configuration parameters

The administrator must define the following data for the local site and remote sites. (See Figures 13-6 and 13-8 to view these fields on the View/Modify Local Site and Add Remote Site screens.)

- Site IDs for the local site and remote sites (each ID must be unique in the network)
- Site names (for the local site and remote sites)
- A dialing plan must be specified for the local site and remote sites. The selected dialing plan must reflect the dialing plan that is used by your organization. A site can be configured with the local switch as ESN, CDP, Hybrid (both ESN and CDP) or as having no numbering plan. (See the following section, "Dialing Plans and Location Codes" for more information.)
- Location codes are used by non-administrative users to distinguish remote mailbox numbers from local ones when composing messages. When local users compose messages to remote users, they must precede the remote user's mailbox with the location code (also referred to as the location prefix). Each site in the network has at least one location code associated with it. Sites participating in a Coordinated Dialing Plan (CDP) may have up to 8 location codes. The way in which you enter location codes depends on the type of dialing plan in effect.
  - When the dialing plan is ESN, you must enter the ESN access code and the ESN prefix. (See the description of ESN dialing plans on page 13-10). An ESN site can have only one ESN prefix.
  - When the dialing plan is CDP , you must specify the CDP steering code(s) (see the description of Coordinated Dialing Plans on page 13-11). A CDP site can have up to eight steering codes.

- When the dialing plan is specified as Hybrid (a combination of ESN and CDP), the first location code you enter must be the ESN prefix. You may then add up to seven CDP steering codes in the remaining fields.
- If no numbering plan is selected (i.e., the dialing plan is set to “None”), or if mailbox numbering does not follow the dialing plan, you must enter the location codes as mailbox prefixes.

- A spoken name may be recorded for remote sites (if the remote site is not part of a Coordinated Dialing Plan with the local site).
- For remote sites, you must specify up to three phone numbers by which to reach the Networking service (the two extra numbers allow contingency routes to be tried if the first number is busy or out of service). See the description of “Network Connection DN1, 2, 3” on page 13-26 for more information.

Dialing Plans and Location Codes

A dialing plan is the set of rules the switch uses to route calls through a public or private phone network.

> ***Note:*** The administrator should define or review the dialing plan for the switch before entering the Meridian Networking data.

Location codes are used when composing messages to users at remote sites. Depending on the dialing plan, this prefix is either already part of the mailbox number (Coordinated Dialing Plan), or must be appended to the beginning of the mailbox number (Uniform Dialing Plan).

You may enter location codes to reflect the same sequence of digits a user would enter when making a phone call to someone at a remote site. Emulation of the existing dialing plan has two advantages:

1 DMS VoiceMail users do not have to remember two sets of numbers.

2 Location code and local mailbox number conflicts can be avoided.

However, you may not want to emulate the dialing plan if the dialing prefixes are lengthy. In this case you may want to enter shorter location codes.

> ***Note:*** The following sections make references to extension DNs and placing calls, however, if mailbox numbering follows the dialing plan (i.e., mailbox numbers are the same as the telephone extensions), these descriptions also apply to mailbox numbers and sending voice messages.

ESN Dialing Plan (Uniform Dialing Plan)

In an ESN (Electronic Switched Network) dialing plan, users placing calls to users at remote sites, must first dial a prefix before dialing the user's extension DN. All users in the network use the same prefix to reach a particular site. (To call a user at the same site, the user placing the call does not need to enter a prefix. The user simply dials the local extension DN.)

An ESN prefix consists of two elements: an *access code*, one or two digits in length, followed by a *routing prefix* of a fixed length (usually 3 digits). (DMS VoiceMail does not require that ESN prefixes be a fixed length, and the length may vary from prefix to prefix.) The ESN access code is used to access ESN routing and may vary from switch to switch. The same access code is usually applied to all switches in the network, but not always. (ESN access codes are similar to trunk access codes and are set independently in each switch.) The routing prefix is a unique number which identifies a particular location within the network.

For example, a user at Site A has the local extension 3000. All users at Site A can reach this user by dialing 3000. A user at Site B has to dial the ESN prefix followed by the extension DN. If the ESN prefix for Site A is 6338 (the "6" is the access code and the ESN number "338" is the routing prefix which specifies the site within the network), users outside of Site A must dial 63383000. This means that a particular extension DN may be repeated in locations having different ESN prefixes. For example, 63383000 and 6463000 have the same local extension DN (3000), yet are unique within the network because of the different ESN prefixes.

It is possible for the leading digit(s) of the local extension to overlap the trailing digit(s) of the ESN prefix. Using the DN 6783000 as an example, 3000 is the local extension DN and the full ESN prefix (the access code plus the routing prefix) is 6783. Here, the “3” is both the last digit in the ESN prefix and the first digit of the local extension. This allows you to expand the range of local numbers available. If overlap were not allowed, local extensions in the range of 3000 to 3999 would not be possible.

Coordinated Dialing Plan

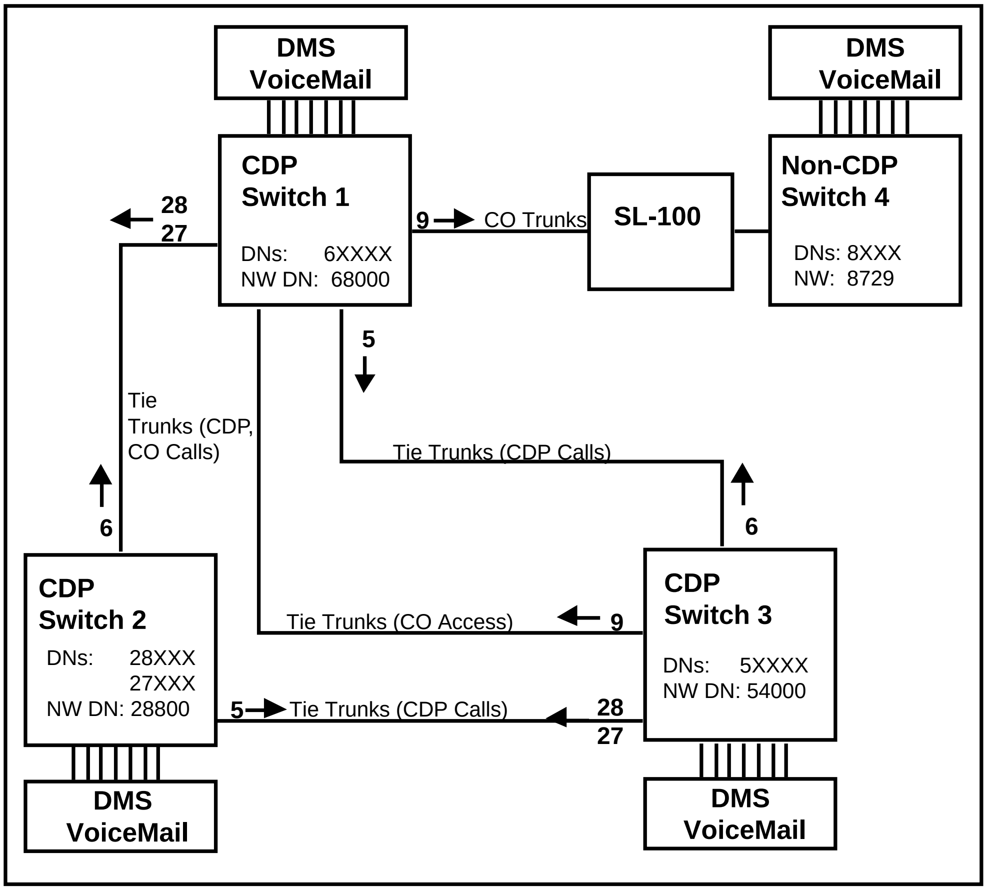
In a Coordinated Dialing Plan (CDP) between two or more switches, a unique dialing number exists for each extension in the network. Unlike ESN, there can be no duplication of extension DNs on different switches. This is due to the fact that the leading digit(s) in an extension DN (known as *steering codes* in a CDP plan) identify a specific site within the network. Calling a user at another site is as easy as dialing a user at your own site - no prefixes or access codes need to be remembered.

For example, in Figure 13-3, the extensions at Site 1 are numbered 60000 to 69999 (the steering code is 6); the extensions at Site 2 are numbered 27000 to 289999 (the steering codes are 27 and 28); the extensions at Site 3 are numbered 50000 to 59999 (the steering code is 5). Regardless of the site at which the user placing the call is, the same extension DN (e.g., 273445) is dialed to reach the user at Site 2. Therefore, users do not need to prefix remote mailbox numbers with additional codes because the first digit (or first two digits) in the DN is the steering code which identifies the site within the network.

No Dialing Plan

When there is no dialing plan, sites may be configured to use different dialing prefixes to reach a specific remote site, however, DMS VoiceMail will not be able to represent the numbering plan. A tie-line is an example of a network with no coordinated dialing plan. In this case a mailbox prefix should be entered to allow users to compose to mailboxes at this remote site, as the mailbox numbering plan will be independent of the dialing plan.

Figure 13-3xxx
Typical CDP Network



Error conditions

Networking maintains a message queue containing a list of messages destined for each remote site. Thresholds based on message priority and message volume control the triggering of connections to remote sites. When the thresholds are reached, Networking attempts to call the remote site.

Networking dials the remote site's connection DN to establish a delivery connection. If the call attempt fails (due to a busy or no answer condition, for example), Networking waits for a preset time before attempting the call again. If three attempts in a row fail then Networking waits for one hour before repeating the three-call-attempt cycle. A SEER is generated indicating that Networking has gone into ERROR state against the remote site. The Networking Status screen reflects this state as well; see "Viewing the networking status" later in this chapter. The Clear Remote Error Sites function is used to clear error conditions for all sites and causes Networking to clear its one-hour delay between the three-call-attempt cycles.

If a subsequent call attempt succeeds in connecting to the remote site, Networking will clear the one hour delay; it does not clear the error conditions for any other remote sites unless a successful connection has been established with these sites or the error condition has been cleared by the administrator using the [Clear Remote Error Sites] softkey. If call attempts to a remote site fail regularly, contact the administrator at that remote site to ensure that the site is operational.

CAUTION
Changing the network data
If you plan on altering the network data fundamentally, such as by changing location codes or by adding or deleting sites/locations, you should do so after hours when users are not logged on. Such changes affect entries in the VSDN table and user directory entries. These entries should be checked and altered accordingly. Otherwise, users may not be able to log on or to compose messages to affected mailboxes. Carefully plan network sites and locations before installation to avoid changing the configuration.

The Network Administration Main Menu

From the Network Administration Main Menu (Figure 13-4) you can choose to perform either Meridian network administration or AMIS network administration. This chapter describes Meridian networking only. AMIS networking is described in the previous chapter. If you do not have the proprietary network feature, this option will not appear and you will go directly to the AMIS network administration screen.

CAUTION
Overnight system audits
If you leave the administrative terminal in any Network Administration menu overnight, important system audits may fail due to a lack of available memory.

Figure 13-4xxx
The Network Administration Main Menu

```
                         Network Administration


        1  Meridian Networking Administration

      * 2  AMIS (Open Access) Networking Administration







 Select an item >

   Exit
```

```
* If AMIS Networking is not installed, this menu will not be displayed.
  Instead, the Network Administration Menu (Figure 13-5) is displayed
  when you select Network Administration from the Main Menu.
```

Procedure 13-1xxx
Using the Network Administration Main Menu

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

To perform AMIS network administration, select AMIS (Open Access) Networking Administration instead and see Chapter 12.

See the next section, “The Network Administration Menu”.

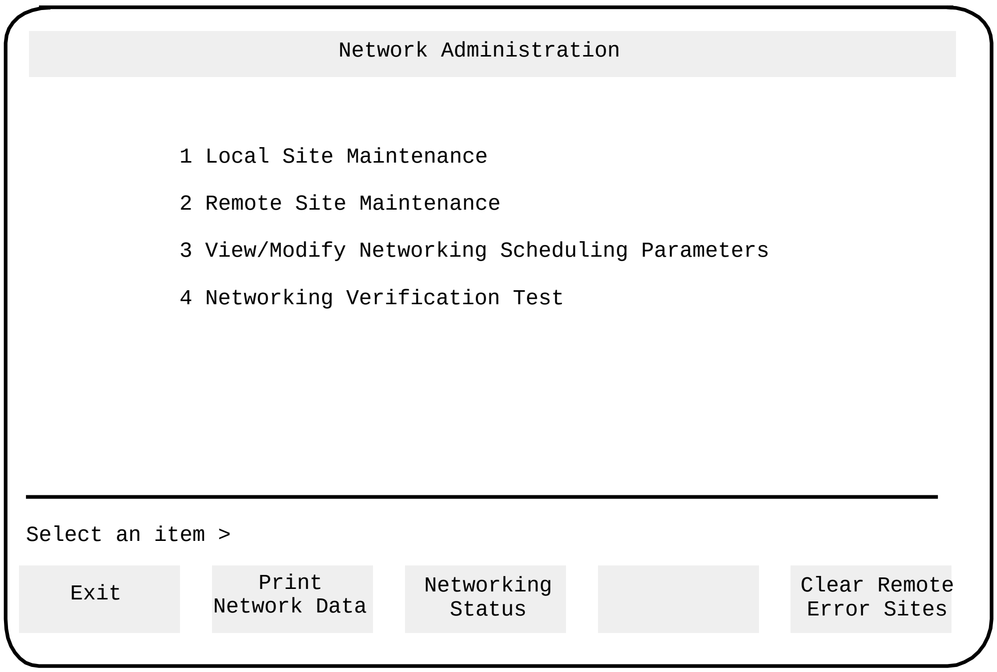
3 To exit this menu, press [Exit].

The Main Menu is displayed.

The Network Administration Menu

Meridian Network administration involves local and remote site configuration, definition of holding times and stale times for urgent, standard and economy messages, and verification of message delivery between sites. The Network Administration menu (Figure 13-4) provides you with functions for adding remote sites, modifying the local site and remote site, and verifying the operational status of connections between sites.

Figure 13-5xxx
The Network Administration Menu



Procedure 13-2xxx
Using the Network Administration menu

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration Menu (Figure 13-5) is displayed.

3 To perform local site maintenance, go to step a. To perform remote site maintenance, go to step b. To view or modify networking scheduling parameters, go to step c. To perform a networking verification test, go to step d. To print network data, go to step e. To view the networking status, go to step f. To clear error conditions in remote sites, go to step g. To exit the Network Administration menu, go to step h.

a. Select Local Site Maintenance.

The View/Modify Local Site screen is displayed (see Figure 13-6).

b. Select Remote Site Maintenance.

The List Remote Sites screen is displayed (see Figure 13-7). From this screen you can add, modify or delete remote sites.

c. Select Networking Scheduling Parameters.

The View/Modify Networking Scheduling Parameters screen is displayed (see Figure 13-11) in which you can configure holding times and stale times for messages of different priorities.

d. Select Networking Verification Test.

The Networking Verification Test screen is displayed (see Figure 13-12). This test allows you to ensure network operation before adding a new site.

e. Press [Print Network Data].

This action prints local and remote site information from the network data base. Ensure that the printer is on-line before making this selection. See page 13-42 for details.

f. Press [Networking Status].

The number of messages queued for transmission are displayed in the Networking Status screen. See Figure 13-13.

g. Press [Clear Remote Error Sites].

Any remote site status that shows error conditions is cleared. View the Networking Status screen to verify the status of the network. See the section "Error Conditions" earlier in this chapter.

h. Press [Exit].

The Network Administration Main Menu is displayed.

Local Site Maintenance

When you select Local Site Maintenance from the Network Administration menu, you are prompted for the Site Number at the bottom of the screen (if you have not yet configured the local site). You will not be able to access the View/Modify Local Site screen until you enter a number.

Note: You cannot delete the local site.

Figure 13-6xxx
The View/Modify Local Site screen

```
                      Network Administration

  Local Site Maintenance - View/Modify Site

  Site Number:   ________________________________________
  Site Name:     ________________________________________

  Message Transfer:Disabled     [Enabled]
  Site is Network Message Center: [No] Yes

  Dialing plan:     ESN  CDP  [Hybrid]  None
  Maximum Number of digits in Local Mailbox:     ________

* ESN Access Codes:      ________    ________
* Number of digits in common between ESN prefix and Local Extension:__

**Number of digits in common between CDP Steering Code and Local
   Extension: _______

  ESN/CDP codes (The ESN code must begin with the first access code above):

  1:_____________ 2: __________
  3:_____________ 4: __________
  5:_____________ 6: __________
  7:_____________ 8: __________
  9:_____________10: __________

  Mailbox Numbering follows Dialing Plan:   No   [Yes]
***  Mailbox Prefixes: _________

    Save          Cancel
```

```
*  Only appears if dialing plan is ESN or Hybrid
** Only appears if dialing plan is CDP or Hybrid
***Only appears if Mailbox Numbering does not follow dialing plan.
```

The following fields are displayed:

- ***Site Number*** - The site number uniquely identifies the local site in the Meridian network. It is entered when you select Local Site Maintenance for the first time. This number (as well as remote site numbers) should be obtained from the network administrator to ensure the number is not already in use. The valid range is from 1 to 499 (although only up to 50 remote sites are supported).

  Note: This value can only be changed by using the “Change Local Site ID” utility. (See Appendix A, “System Administration Tools”. Before using this utility, you will have to create a dummy remote site as this utility only accepts an existing remote ID as the new local site ID. If you change the local site ID after users have been set up, the system will not recognize their mailbox numbers due to invalid site IDs.
- ***Site Name*** - This field is mandatory. The site name is usually the same as your organization’s name. The field can contain up to 32 alphanumeric characters. There is no default.
- ***Message Transfer*** - This field allows you to enable or disable networking at the local site. This field must be enabled for users at the local site to send messages to remote sites. The default is “Enabled”.
- ***Dialing Plan*** - The selection you make must reflect your organization’s dialing plan. The default is “Hybrid”. (See “Dialing Plans and Location Codes” earlier in this chapter.)
- ***Maximum Number of Digits in Local Mailbox*** - Enter the maximum number of digits allowed in mailbox numbers at the local site. If you are unsure, enter the maximum allowable value of 8. The default is “4”.

  Make sure that this value is not less than any existing mailbox lengths. If any mailbox numbers are longer than the value specified here, these users will not be able to gain access to their mailboxes.

  Note: If you try to change this field, you will see the following message: “Warning: Reducing Maximum Digits may affect some users”.
- ***ESN Access Codes*** - This field is mandatory if the dialing plan is ESN. This code is used to access the ESN network from this site’s switch. You may enter two different ESN access codes of 1 or 2 digits each. A typical access code is “6” or “9”. Check your dialing plan for the ESN access code for the local site’s switch.
- ***Number of Digits in Common between ESN Prefix and Local Extension:*** - This field appears only when the dialing plan is ESN or Hybrid. This number indicates the number of digits in the ESN prefix that overlap with the extensions that are local to the remote site. For example, your local extensions are 5 digits long and all begin with “8”. Your ESN prefix is “338”. If you enter “0” (no overlap) in this field, users at remote sites will have to enter an 8-digit DN when addressing messages to your site (such as 33883000). If you enter “1” in this field,

indicating that the last digit of the prefix and the first digit of the extension overlap, remote users will specify a 7-digit address (3383000). The selection you make here must conform with your local site's dialing plan (if there are digits in common between the prefix and local extensions, they may or may not overlap.)

Note 1: To enter a non-zero value in this field, the mailbox numbers of all users you enter for this site must begin with the overlap digit(s) of the ESN prefix. For example, if the last digit of the ESN prefix ("8") overlaps with the local extension, then all mailboxes at this site must begin with "8".

Note 2: You must enter a value in this field. If left blank, it will revert to the previous value. If there are no numbers in common between the ESN prefix and the local extensions, use the default setting of "0".

- ***Number of Digits in common between CDP Steering Code and Local Extension*** - This field only appears if the dialing plan is CDP or Hybrid. This number indicates the number of digits in the CDP steering code that overlap with the local extension. These codes need not overlap. Specify the length of the CDP code in this field. Refer to your organization's dialing plan.

  Note: You must enter a value in this field. If left blank, it will revert to the previous value. If there are no numbers in common between the CDP code and the local extensions, enter a value of "0".

- ***ESN/CDP Codes*** - These codes identify the local site within the network according to a CDP or ESN numbering plan. They are used by users at remote sites to send messages to users at the local site. The code must be unique within the ESN or CDP network. Consult with your network administrator to verify that the code is not being used by another site.

  Note: It is not necessary to enter ESN prefixes and/or CDP steering codes for the local site. (They must, however, be specified for remote sites.) This is because the local site will be defined as a remote site in the database of the other sites that are part of the DMS VoiceMail network. This means that the ESN prefix or CDP steering code for the local site will be defined at all of the other network sites (in their Remote Site Maintenance screens). This is how users at other sites will be able to identify the local site. (You can enter the ESN prefix/CDP code for completeness if you wish. If you choose to do this, users at the local site will be able to compose messages to other users at their own site using the network format. Note that for CDP dialing plans, the steering codes are usually already fully incorporated within the mailbox numbers.)

  The first field is intended for the ESN prefix. In a CDP-only dialing plan up to nine steering codes may be entered for a site. For example, if either 35xxx or 36xxx can be used to reach extension xxx at the local site, enter 35 and 36 as the steering codes.

- ***Mailbox Numbering follows Dialing Plan*** - This field allows you to select whether or not mailbox numbering will emulate the dialing plan (telephone extensions). If you answer “No”, the following fields, *Mailbox prefixes*, appears. The default is “Yes”.
- ***Mailbox Prefixes*** - These prefixes identify the local site within the network when mailbox numbering does not follow the dialing plan. This prefix does not have any overlap with local mailbox numbers and is independent of the ESN prefix and CDP steering codes. Make sure that the numbers you enter do not conflict with other other network data.

Procedure 13-3xxx
Configuring the local site

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration Menu (Figure 13-5) is displayed.

3 Select Local Site Maintenance.

The View/Modify Local Site screen is displayed (see Figure 13-6).

4 Fill in the fields as described in the preceding pages.

5 To save the configuration, go to step a. To exit the screen without saving your changes, go to step b.

a. Press [Save].

Any changes that you have made are saved. The Network Administration menu is displayed.

b. Press [Cancel].

Any changes that you have made are discarded and the Network Administration screen is displayed.

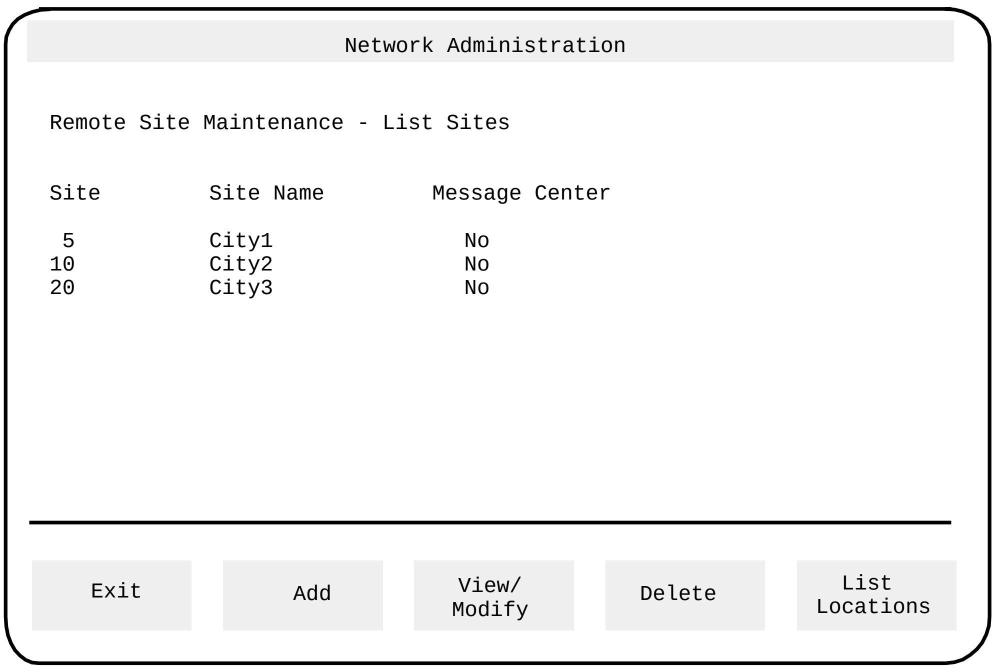
Remote Site Maintenance

When you select Remote Site Maintenance from the Network Administration menu, the List Remote Sites screen is displayed. Through various screens you can add, modify and delete remote sites.

Listing remote sites

The List Remote Sites screen (Figure 13-7) lists all of the remote sites that are part of the DMS VoiceMail network. The softkeys displayed on this screen allow you to add new remote sites to the network, select an existing site in order to view, modify or delete it.

Figure 13-7xxx
The List Remote Sites screen



The following read-only fields are displayed:

- ***Site*** - The Site ID for the remote site.
- ***Site Name*** - The name corresponding to the remote site.
- ***Message Center*** - This field indicates whether or not the remote site is also an Network Message Service (NMS) site. NMS is not available with DMS VoiceMail or with Meridian Mail on the MSM.

Procedure 13-4xxx
Listing remote sites

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration menu is displayed.

3 Select Remote Site Maintenance.

The List Remote Sites screen (Figure 13-7) is displayed.

4 To add a remote site, go to step a. To view or modify an existing remote site, go to step b. To delete a remote site, go to step c. To exit this screen, go to step d.

a. Press [Add].

See the following section, “Adding remote sites”.

b. Use the up/down cursor keys to select the site to want to view or modify and press [View/Modify].

See the section “Viewing and modifying remote sites” for details.

c. Use the up/down cursor keys to select the site to want to delete and press [Delete].

See the section “Deleting remote sites” for details.

d. Press [Exit].

The Meridian Network Administration menu is displayed.

5 To save the configuration, go to step a. To exit the screen without saving your changes, go to step b.

a. Press [Save].

Any changes that you have made are saved. The Network Administration menu is displayed.

b. Press [Cancel].

Any changes that you have made are discarded and the Network Administration screen is displayed.

Adding remote sites

The Add Remote Site screen (Figure 13-8) allows you to define new remote sites in the DMS VoiceMail network. When you press the [Add] softkey on the List Remote Sites screen, you are prompted for a remote site ID. After you enter an ID and press <Return>, the Add Remote Site screen is displayed.

If AMIS is installed at the local site, remote AMIS sites can also be added to the DMS VoiceMail network from the Add Remote Site screen. Certain fields need to be configured somewhat differently than for normal remote sites: these are the *Message Transfer Protocol* field and the *Networking Connection* fields. See the descriptions of these fields for more details.

Note: Before adding a remote site to the DMS VoiceMail network you should perform a verification test to ensure proper operation of the networking service between the local site and the remote site. The networking verification test is described on page 13-38.

Figure 13-8xxx
The Add Remote Site screen

```
                          Network Administration

  Remote Site Maintenance - Add Site

  Site Number:   _______________________
  Site Name:     _______________________

  Message Transfer Protocol:      [Meridian]  AMIS
  Message Transfer:               Disabled  [Enabled]

  Networking Connection:
    DN 1:  _______________________________
    DN 2:  _______________________________
    DN 3:  _______________________________

  Password
    Initiating: _______________________
    Responding: _______________________

  Dialing plan:    ESN  CDP  [Hybrid]  None
  Maximum Number of Digits in Local Mailbox:____________

* ESN Access Codes:      _________   __________
* Number of Digits in common between ESN Prefix and Local Extension ____

**Number of Digits in common between CDP Steering Code and Local
  Extension:________

**ESN/CDP codes (The ESN code must begin with n)::

  ESN/CDP codes (The ESN code must begin with the first access code
  above):

  1:_____________    2:__________
  3:_____________    4:__________
  5:_____________    6:__________
  7:_____________    8:__________
  9:_____________   10:__________

   Mailbox Numbering follows Dialing Plan:     No   [Yes]
 ***Dial Prefix:        __________
****Mailbox Prefixes: __________  __________
________________________________________________________________________
*Site Name recorded (voice):  No
    Save         Cancel                     Voice*
```

```
*    Only appears if dialing plan is ESN or Hybrid
 ** Only appears if dialing plan is CDP or Hybrid
*** Only appears if dialing plan is none.
****Only appears if dialing plan is none or if Mailbox Numbering does not
     follow the dialing plan.
```

Note: The “n” in the *ESN/CDP codes* field represents the ESN access code that is configured in the Local Site Maintenance screen. If two access codes are defined in that screen, the one entered in the first field is displayed here.

The following fields are displayed on the Add Remote Site screen:

- ***Site Number*** - This field is mandatory. The site number uniquely identifies the remote site in the Meridian network. Site numbers should be obtained from the network administrator to ensure the number is not already in use. The valid range is from 1 to 500.
- ***Site Name*** - This field is mandatory. The site name should uniquely identify the remote site. The field can contain up to 32 alphanumeric characters. There is no default.
- ***Message Transfer Protocol*** **-** This field is displayed only if AMIS networking is installed at the local site. The default is “Meridian”, however, you must set this field to “AMIS”.
- ***Message Transfer*** - This field must be set to “Enabled” for local users to send messages to the remote site. Select “Disabled” if you must temporarily disable message delivery to the remote site. The default is “Enabled”.
- ***Networking Connection DN 1, DN 2, DN 3*** - These are the telephone numbers that are used to establish a connection to the Networking service at the remote site. Because DNs are used in the given order, enter the least costly DN as DN 1, and the most costly DN as DN 3. Numbers are between 3 and 30 digits long. They may include the digits 0 to 9 and special symbols * and #, where * inserts a 3-second pause in the sending of digits, and # indicates end-of-dialing. A minimum of one DN must be defined.

  Enter either a private or public system access number as DN1. Private system access numbers are entered in the format:

  0##n#

  where n is a number up to 30 digits in length (the initial 0 indicates a private network and must be followed by two pound signs).

  Public system access numbers are entered in the format:

  ccc#aaa#nnnnnnn#

  where ccc is the country code, aaa is the area code and nnnnnnn is the local number. Use # to separate the codes and terminate the number. (You do not have to enter the access code, such as “9”.”

  If you enter a private access number as DN1, you must enter a public access number as DN2.
- ***Password (Initiating and Responding)*** - Not applicable. Y ou may leave these fields blank.

- ***Dialing Plan*** - The selection you make must reflect your organization's dialing plan. The default is "Hybrid". (See "Dialing Plans and Location Codes" earlier in this chapter.) Note that in a CDP dialing plan, remote sites are transparent to voice messaging users because no special prefixes are required to dial out to them. Therefore if this site is part of a CDP dialing plan only, the [Voice] key is not displayed as a personal verification is not required.

  Note: If you change a site from Hybrid to CDP, the personal verification is removed. If you change the site back to Hybrid, you will have to re-record the verification.

- ***Maximum Number of Digits in Local Mailbox*** - Enter the maximum number of digits allowed in mailbox numbers at the remote site. If you are unsure, enter the maximum allowable value of 8. The default is "4".

  Make sure that this value is not less than any existing mailbox lengths. If any mailbox numbers are longer than the value specified here, these users will not be able to gain access to their mailboxes.

  Note: If you try to change this field, you will see the following message: "Warning: Reducing Maximum Digits may affect some users".

- ***ESN Access Codes*** - This field is mandatory if the dialing plan is ESN. This code is used to access the ESN network from this site's switch. You may enter two different ESN access codes of 1 or 2 digits each. A typical access code is "6" or "9". Check your dialing plan for the ESN access code for the local site's switch.

  Note: If one of your ESN access codes is "9", this will create a conflict with the default broadcast mailbox number (999). You will have to change the broadcast mailbox number in order to avoid this conflict.

- ***Number of Digits in Common between ESN Prefix and Local Extension:*** - This field appears only when the dialing plan is ESN or Hybrid. This number indicates the number of digits in the ESN prefix that overlap with the extensions that are local to the remote site. For example, your local extensions are 5 digits long and all begin with "8". Your ESN prefix is "338". If you enter "0" (no overlap) in this field, users at remote sites will have to enter an 8-digit DN when addressing messages to your site (such as 33883000). If you enter "1" in this field, indicating that the last digit of the prefix and the first digit of the extension overlap, remote users will specify a 7-digit address (3383000). The selection you make here must conform with your local site's dialing plan (if there are digits in common between the prefix and local extensions, they may or may not overlap.)

Note 1: To enter a non-zero value in this field, the mailbox numbers of all users you enter for this site must begin with the overlap digit(s) of the ESN prefix. For example, if the last digit of the ESN prefix (“8”) overlaps with the local extension, then all mailboxes at this site must begin with “8”.

Note 2: You must enter a value in this field. If left blank, it will revert to the previous value. If there are no numbers in common between the ESN prefix and the local extensions, use the default setting of “0”.

- ***Number of Digits in common between CDP Steering Code and Local Extension*** - This field only appears if the dialing plan is CDP or Hybrid. This number indicates the number of digits in the CDP steering code that overlap with the extensions that are local to the remote site. These codes need not overlap. Specify the length of the CDP code in this field. Refer to your organization’s dialing plan.

  Note: You must enter a value in this field. If left blank, it will revert to the previous value. If there are no numbers in common between the CDP code and the local extensions, enter a value of “0”.

- ***ESN/CDP Codes*** - The ESN prefix and CDP steering codes are location prefixes that identify the remote site within the network and must therefore be unique within the ESN or CDP network. Check with the administrator at the remote site to determine the ESN prefix or CDP steering codes that apply.

  When a user at the local site sends a voice message to a user at the remote site that uses an ESN numbering plan, he or she must precede the remote mailbox number with the ESN prefix entered here. If the remote site is part of a CDP network, the CDP steering code is already part of the mailbox number as far as users are concerned. (Even though this is the case, the CDP steering codes must still be defined here because the system needs to be able to distinguish the steering code from the mailbox number.)

  The first field is intended for the ESN prefix. (Only one ESN prefix can be entered.) This prefix must begin with the one- or two-digit access code that is defined in the Local Site Maintenance screen. The actual access code is displayed in the field title to remind you. For example, if the ESN access code is “6” and the ESN prefix is “344”, enter “6344” in this field.

  If the dialing plan for the remote site is “Hybrid”, enter the ESN prefix in the first field and up to nine CDP steering codes.

  In a CDP-only dialing plan up to ten steering codes may be entered for a site. For example, if either 35xxx or 36xxx can be used to reach extension xxx at the remote site, enter 35 and 36 as the steering codes.

- ***Mailbox Numbering Follows Dialing Plan*** - In this field, specify whether or not mailbox numbering emulates the dialing plan (telephone extensions) at the remote site. If it does not, the following field, *Mailbox prefixes*, is displayed and you must enter the location codes as mailbox prefixes.
- ***Mailbox Prefixes*** - These prefixes function as location codes and are used to identify the remote site within the network when there is no dialing plan or if mailbox numbering does not follow the dialing plan at the remote site. These prefixes do not have any overlap with local mailbox numbers and are independent of the ESN location prefix and CDP steering codes. Make certain that these prefixes do not conflict with other network data.
- ***Dial Prefix*** - This field is displayed only if the dialing plan is “None”. (However, the mailbox numbering at the remote site must follow the extension numbering.) This is an optional prefix which allows users at the local site to use the Call Sender feature to automatically dial the number of a user who has sent a message from the remote site you are defining.

  If an ESN or CDP numbering plan is present, the system figures out the Call Sender number using the specified dialing plan and the mailbox number and there is no need for a dial prefix. Note that if mailbox numbering at the remote site does not follow local extension numbering, you will *not* be able to specify a dial prefix, and therefore, users at the local site will not be able to use the Call Sender feature to that particular remote site.
- ***Site Name Recorded (Voice)*** - This field applies only to ESN and Hybrid dialing plans. If set to “Yes’, the administrator may record a spoken name for the site using the [Voice] softkey. If a spoken name is recorded for a site, voice messaging users hear the site name followed by the local mailbox digits. If there is no spoken name, users hear the ESN location prefix spoken out. The default is “No”.

Procedure 13-5xxx
Adding remote sites

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration menu is displayed.

3 Select Remote Site Maintenance.

The List Remote Sites screen (Figure 13-7) is displayed.

4 Press [Add].

The Add Remote Site screen is displayed. See Figure 13-8.

5 Fill in the fields as required. See the field descriptions on the preceding pages.

6 For ESN or Hybrid dialing plans, record a name for the remote site by moving the cursor to the *Site name recorded* field and pressing [Voice] while the cursor is on the field.

See the section "Recording site names using the [Voice] softkey" later in this chapter for more information.

7 To save the remote site configuration, go to step a. To exit the screen without saving your changes, go to step b.

a. Press [Save].

Any changes that you have made are saved. You are prompted to enter the next remote site ID. If you have more sites to add, enter the ID and press <Return>. If you do not have more sites to add, use the [Cancel] softkey.

b. Press [Cancel].

Any changes that you have made are discarded and the List Remote Sites screen is displayed.

Viewing and modifying remote sites

Once you have added remote sites to the network, you can alter their characteristics by accessing View/Modify Remote Site screen. To display the screen, access the List Remote Sites screen, move the cursor to the site you want to modify and press the [View/Modify] softkey. This screen is identical to the Add Remote Site screen. See the previous section, “Adding remote sites”, for descriptions of the fields and the actions that are possible.

Procedure 13-6xxx
Viewing and modifying remote sites

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration menu is displayed.

3 Select Remote Site Maintenance.

The List Remote Sites screen (Figure 13-7) is displayed.

4 Press [View/Modify].

The View/Modify Remote Site screen is displayed. It is identical to the Add Remote Site screen.

5 Fill in the fields as required. See the field descriptions on the preceding pages.

6 For ESN or Hybrid dialing plans, record a name for the remote site by moving the cursor to the *Site name recorded* field and pressing [Voice] while the cursor is on the field.

See the section “Recording site names using the [Voice] softkey” later in this chapter for more information.

7 To save the remote site configuration, go to step a. To exit the screen without saving your changes, go to step b.

a. Press [Save].

Any changes that you have made are saved and the List Remote Sites screen is displayed.

b. Press [Cancel].

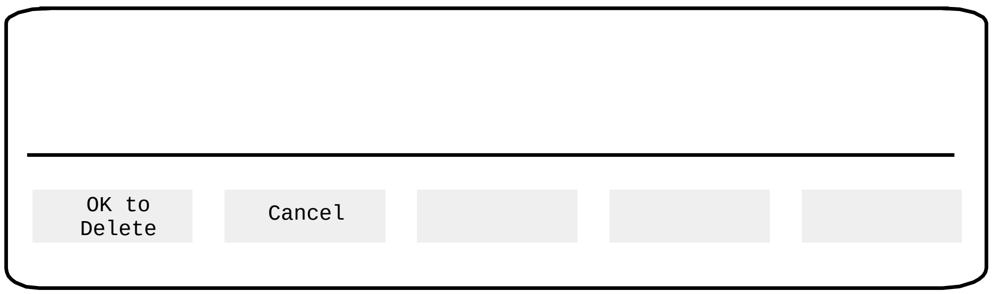
Any changes that you have made are discarded and the List Remote Sites screen is displayed.

Deleting remote sites

Remote sites are removed from the network database from the List Remote Sites screen. When you select a site and then press the [Delete] softkey, a new set of softkeys is displayed.

Messages that have been sent but not delivered to a remote site that has been deleted are returned with a non-delivery notification (NDN).

Figure 13-9xxx
The Delete Remote Site softkeys



Procedure 13-7xxx
Deleting remote sites

Starting point: The List Remote Sites screen.

1 Use the cursor keys to highlight the remote site you want to delete.

2 Press the <Spacebar> to select the site.

3 Press the [Delete] softkey.

The softkeys shown in Figure 13-9 are displayed.

4 Choose step 4a to delete the location, or 4b to cancel the delete operation and return to the List Remote Sites screen.

a. Use [OK to Delete].

The system purges the site. You are prompted for another site number.

To delete another site, enter the site number and press <Return>.

b. Use [Cancel].

The site is not deleted and the List Remote Sites screen is re-displayed.

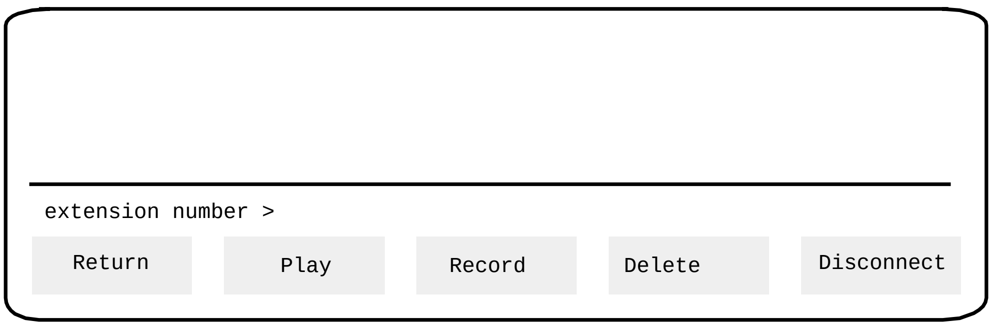
Recording site names using the [Voice] softkey

The [Voice] softkey is used to provide a new set of softkeys for recording, playing and deleting remote site names. When a verification is recorded for a site, the *Site Name Recorded (Voice)* field is automatically set to “Yes” in the View/Modify Remote Site screen. Within Networking Administration, the [Voice] softkey is available on the Add Remote Site and View/Modify Remote Site screens.

> ***Note 1:*** If a site is part of a CDP dialing plan only, the [Voice] softkey is not displayed and you cannot record a site name. If a verification is recorded for a Hybrid site which is then changed to CDP-only, the verification is removed. If the site is changed back to Hybrid, the verification must be re-recorded.

> ***Note 2:*** A telephone set is required to record the site name. Ensure that a phone set is available near the administration terminal where you are working.

Figure 13-10xxx
Recording softkeys



Procedure 13-8xxx
Recording site names

Starting point: The Add Remote Site or View/Modify Remote Site screen.

1 Press the [Voice] softkey.

The row of softkeys changes to display a set of recording softkeys (Figure 13-10). You are prompted to enter an extension number.

2 Enter the extension number of the phone set you are going to use to record a spoken name.

3 Use step 3a to record a new spoken name, step 3b to play an existing verification, step 3c to delete a verification, step 3d to disconnect the extension number or step 3e to return to the original set of softkeys.

a. Pick up the handset of the phone and then press [Record]. Wait for the beep and record the site name. When you press the [Record] softkey, a new [Stop] softkey appears. Press the [Stop] softkey to stop the recording when you are done.

b. Pick up the handset of the phone and press [Play] softkey.

If a verification is recorded, it will be played over the phone.

c. Pick up the handset of the phone and press [Delete] softkey.

   If a verification was recorded, it will be deleted and a prompt will be played advising you that the recording was deleted.

d. Press [Disconnect]

   The extension you are using to record the verification will be disconnected.

e. Press [Return] to remove the voice softkey set and return to the original screen and softkeys.

Networking Scheduling Parameters

These parameters control how long messages are kept before they are sent to remote sites, how long unsuccessfully delivered messages are kept before they are purged, and how often the system checks for queued messages and sets up connections for delivery to remote sites. The Networking Scheduling Parameters screen is accessed by selecting item <3>, View/Modify Networking Scheduling Parameters, from the Network Administration menu (Figure 13-5). Thresholds (holding times and stale times) and message priorities (urgent, standard and economy) are described near the beginning of this chapter.

Figure 13-11xxx
The Networking Scheduling Parameters screen

```
                         Network Administration


Networking Scheduling Parameters

     Economy Class Initiation Time (hh:mm):     18:00
     Economy Class Stale Time      (hh:mm):     06:00

     Standard Class Holding Time   (hh:mm):     03:00
     Standard Class Stale Time:    (hh:mm):     09:00

     Urgent Class Holding Time     (hh:mm):     00:30
     Urgent Class Stale Time       (hh:mm):     01:30

     Batch Threshold:                           3
     Wakeup Interval (minutes):                 3
     Networking call maximum:                   4

     Save          Cancel
```

The following fields are displayed:

- ***Economy Class Initiation Time*** - This is the time at which delivery of economy messages begins. Economy messages, unlike urgent and standard message, are delivered only once a day at a particular time. Enter the time in hours and minutes in the range 00:00 to 23:59. The default is “18:00”.

- ***Economy Class Stale Time*** - The value entered in this field determines the maximum retention time for messages tagged as economy. When this threshold is reached, a non-delivery notice is sent to the originator and the message has to be composed and sent again. Enter the time in hours and minutes in the range 03:00 to 99:59. The default is “06:00”.
- ***Standard Class Holding Time*** - The value entered in this field determines the length of time that a standard priority message is retained before the system attempts to send it. A message may be transferred before this holding time expires if a connection is established for another reason, such as delivering urgent messages. Enter the time in hours and minutes in the range 00:00 to 33:20. The default is “03:00”.
- ***Standard Class Stale Time*** - The value entered in this field specifies the maximum retention time for messages tagged as standard. If a message is not delivered before this time, a non-delivery notice is sent to the originator. These messages have to be composed and sent again. The time is entered in hours and minutes. This value must be at least three times the standard class holding time (see the note below). The maximum value you can enter in this field is 99:59. The default is “09:00”.

  Note: The standard and urgent stale times must be at least three times greater than the holding time. (Enter the holding time for each class first.) The holding time you enter affects the range of stale times that you can enter. For example, if you entered a standard holding time of 00:20, your standard stale time would have to be in the range 01:00 to 99:59. If you entered an urgent holding time of 05:00, your urgent stale time would have to be in the range 15:00 to 99:59.
- ***Urgent Class Holding Time*** - The value entered in this field determines the length of time that an urgent priority message is retained before the system attempts to send it. A message may be transferred before this holding time expires if a connection is established for another reason. The time is entered in hours and minutes and must be in the range 00:00 to 33:20. The default is “00:30”.
- ***Urgent Class Stale Time*** - The value entered in this field is the maximum retention time for messages tagged as urgent. If a message is not delivered before this time, a non-delivery notice is sent to the originator. These messages have to be composed and sent again. The time is entered in hours and minutes and must be at least three times the urgent class holding time. The maximum value you can enter is 99:59. The default is “01:30”.
- ***Batch Threshold*** - The value entered in this field specifies the total number of standard and urgent messages that can accumulate for a given site before delivery commences. The maximum is 99 and the default is “20”.

- ***Wakeup Interval (minutes)*** - This is the periodic interval at which the networking software checks for messages that are waiting to be sent and sets up the connections required to send those messages. The default is “5”.
- ***Networking Call Maximum*** - The maximum number of simultaneous outgoing networking calls permitted. If this maximum is reached, no new outgoing sessions will be attempted. This prevents Meridian networking from using too many resources and interfering with the effective functioning of other DMS VoiceMail services. This limit applies to both DMS VoiceMail networking calls and AMIS networking calls. The valid range is from 1 to 99. The default is “4”.

Procedure 13-9xxx
Setting networking scheduling parameters

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration menu is displayed.

3 Select View/Modify Networking Scheduling Parameters.

The View/Modify Networking Scheduling Parameters screen (Figure 13-11) is displayed.

4 Customize the parameters as required. See the field descriptions on the preceding pages for more information.

5 To save the new parameters, go to step a. To exit the screen without saving your changes, go to step b.

a. Press [Save].

Any changes that you have made are saved and the Network Administration menu is displayed.

b. Press [Cancel].

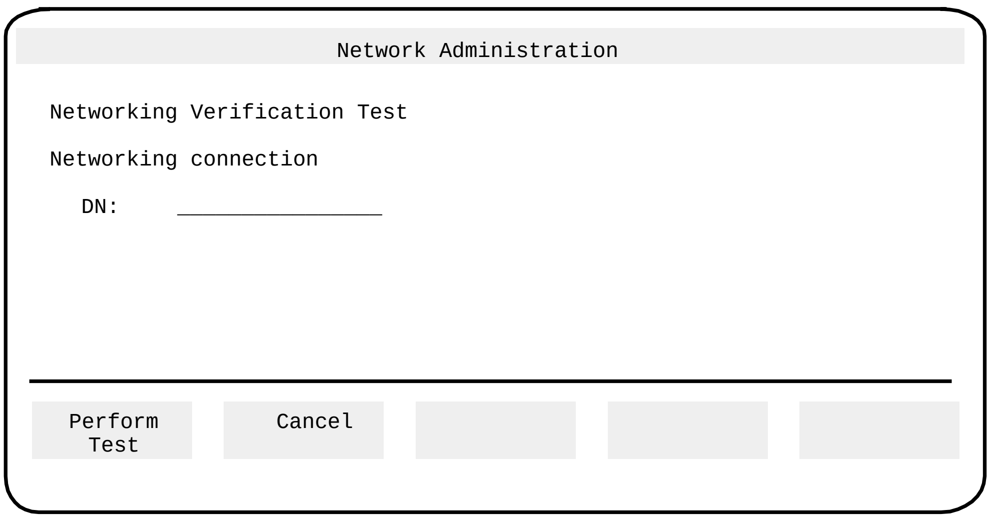
Any changes that you have made are discarded and the Network Administration menu is displayed.

Networking Verification Test

Before adding a remote site to the Meridian network you should perform a verification test to ensure proper operation of the networking service between the local site and the remote site. This is a loopback test which verifies network message delivery between systems. Instead of delivering a voice message to a remote system, this test delivers the message back to your system.

> ***Note:*** This test requires one outbound trunk, one inbound trunk, and two 2500 set lines - one for outgoing and one for the incoming networking call. Sites with only one modem cannot perform the test.

Figure 13-12xxx
The Networking Verification Test screen



Before running the networking verification test, also test the following:

1 Call the Networking DN (that is defined in the VSDN table). You should received the modem tone.

2 Call the modem DN. You should get Ring No Answer.

Procedure 13-10xxx
Testing the networking service

Starting point: The Network Administration menu, item <4> selected.

The verification feature ensures that at least two data lines are configured. If not, an error message appears and you are returned to the Network Administration menu. If the lines are configured the Networking Verification Test screen is displayed (Figure 13-12).

1 Enter the DN of the loopback connection.

The directory number cannot be a local extension number. It must be external to the switch. The loopback DN must dial out of the switch and re-enter it. It should therefore include the outbound trunk access code (such as “9”), and a number that accesses an incoming trunk and terminates on the Networking service. Typically the loopback DN is the phone number that other sites dial to reach the local site.

2 Choose step 2a to carry out the test, or 2b to exit.

a. Use [Perform Test].

A message is automatically created and submitted to the Networking service for delivery. The Networking call is placed and the Networking service accepts the incoming call.

If the connection fails, the loopback feature retries the number. If failure persists the loopback feature abandons the test after 10 minutes and informs you of this.

If the call is successfully made, the message is transferred over the connection. Upon transfer completion the loopback feature verifies that the message has been received and is valid. You are informed of the test results. If the loopback fails, consult the SEER log for the specific reason.

The loopback test causes any active transfers with other sites to be terminated. This is normally not a problem because the site verification test is run before the site joins the live network. However, this will delay message delivery if you use loopback to perform diagnostics on a live site that is experiencing problems (normal operation resumes once the tests are completed).

Use [Cancel] at any time to stop the test. To restart the test, go to 2a. To exit, go to 2b.

b. Use [Cancel].

You are returned to the Network Administration menu.

Viewing the networking status

The [Networking Status] softkey on the Network Administration menu allows you to view the Networking Status screen which lists the activity status of the networking service at each site in the network and the number of economy, standard and urgent messages that are queued for transmission to remote sites. This screen is not dynamic (i.e., it does not automatically update while it is displayed). You can, however, use the [Update] softkey to refresh the screen and update the status while you are viewing it.

Figure 13-13xxx
The Networking Status Screen

```
                           Network Administration

Networking Status

Total Number of Remote Sites:    nnn

Site    Site Name      Status      #Economy       #Standard    #Urgent

100     DownTown       Ready       3              11           1
105     UpTown         Error       0              1            0
111     Jackson        Disabled    13             5            1
113     Riverview      Idle        0              0            0
119     Wickham        Active      15             6            0

   Exit                                                            Update
```

The following read-only fields are displayed:

- ***Total Number of Remote Sites* -** This field displays the number of remote sites that are currently configured in the Meridian network database.

- ***Site* -** The remote site number. The number of each configured remote site is listed in this field.
- ***Site Name* -** The site name corresponding to the site number.
- ***Status* -** The current status of the networking service at that remote site. The status may be one of the following:
  - ***Active* -** indicates that the networking service is operational and is currently sending or receiving messages.
  - ***Idle* -** indicates that the networking service is operational but that there are no networking messages waiting to be delivered.

- ***Ready*** **-** indicates that networking messages are queued and are ready to be transmitted.
- ***Error*** **-** indicates that the networking service failed to deliver a message to the specified remote site (after three attempts).
- ***Disabled*** **-** indicates that message transfer has been disabled for the site. This usually indicates that a site has been taken down for maintenance.

For each site, the number of messages (economy, standard and urgent) are displayed. If the status is idle, the following three fields will display 0 (zero) since no messages are in the queue. For all other states, the following fields will show the number of messages that are currently in the queue.

- ***#Economy*** **-** The number of economy messages that are queued at the specified site for transmission to other sites.
- ***#Standard*** **-** The number of standard messages that are queued at the specified site for transmission to other sites.
- ***#Urgent*** **-** The number of urgent messages that are queued at the specified site for transmission to other sites.

Procedure 13-11xxx
Viewing the networking status

Starting Point: The Main Menu

1 Select Network Administration.

2 Select Meridian Networking Administration.

The Network Administration menu is displayed.

3 Press [Networking Status].

4 To update the status, go to step 4a. To exit the screen, go to step 4b.

a. Press [Update].

The screen is updated to show the current status.

b. Press [Exit].

The Network Administration menu is displayed.

Printing network data

The [Print Network Data] softkey on the Network Administration menu allows you to print the local and remote site information in the database.

Procedure 13-12xxx
Printing local and remote site information

Starting point: The Network Administration menu.

1 Ensure that the printer is on-line.

2 Use [Print Network Data]

The network data (local and remote site information) is printed. The menu prompt reappears when the printing is completed.

3 Use [Cancel] to abort printing at any time. The printing will stop at the end of the current site entry.

Clearing errors at remote sites

The [Clear Remote Error Sites] softkey on the Network Administration menu can be used to reset error conditions on the Network.

Procedure 13-13xxx
Resetting error conditions

Starting point: The Network Administration menu.

1 Press the [Clear Remote Error Sites] softkey.

Any remote site status that shows error conditions is cleared. Use the Networking Status screen (Figure 13-13) to verify the status of the network.

Appendix A: System Administration Tools

Introduction

Note: Some of the utilities described in this document are feature-dependent and may not be installed on your system.

The TOOLS level provides access to some of the following system management utilities:

- ***Move User*** - allows you to move users from one volume to another, one at a time.
- ***Modify Hardware*** - is used to modify the hardware database.
- ***Set Silence Compression*** - compresses out/leaves in recorded silence.
- ***Control Volume*** - allows you to control volume on DMS V oiceMail voice sessions.
- ***Update MWI*** - updates Message Waiting Indicators (MWIs) on telephone sets after the CSE is rebooted.
- ***Block DMS VoiceMail*** - allows you to specify whether or not access to DMS VoiceMail should be blocked in the case of a serious disk failure.
- ***Find Users*** - allows you to find users in the system.
- ***Audit all Volumes -*** allows you to free up data blocks on all volumes in the system.
- ***Synchronize Disks -*** provides you with a set of commands for maintaining the shadowed disks. (This is only available on certain hardware platforms, as described in the chapter “Synchronize Disks”.)
- ***Other*** - consists of other system/feature dependent options.

The following utilities are feature-dependent and will not be displayed if the necessary feature is not installed. (The available utilities are displayed when you select “Other” from the Tools menu.)

- ***Change local site ID*** - allows you to transfer voice prompt files between DMS VoiceMail Systems. This is used for networked systems.
- ***Voice Prompt Transfer Tool*** - allows you to transfer voice prompt files between DMS VoiceMail Systems.

- ***ACCESS Toolkit Diagnostic Tool*** - allows you to diagnose and monitor system activity related to Meridian ACCESS.
- ***Configure UATs*** - allows you to configure the user administration terminals in your system.

CAUTION
Rebooting the system
After using a TOOLS-level utility to modify the hardware database, always reboot the system for the changes to take effect.

The Logon screen

The Logon screen (Figure A-1 or Figure A-2) appears when the administrative console is idle. When the system is installed, the default administration password is "adminpwd" and the password used to access the Tools menu is "tools". To ensure system security, change the administrator password as soon as possible. An unsuccessful logon attempt is automatically recorded in the system log file. As a security precaution, the system forces a ten minute delay after a third unsuccessful attempt to log on, before a further logon attempt will be accepted. Only your Northern Telecom representative has the requisite privileges to gain access to the system during the lockout period. To log on to the system and gain access to the Tools, use Procedure A-1, described on the following page.

Figure A-1xxx
The DMS VoiceMail Logon screen

```
                                  Logon/Status                       MM8


                           ______________________________________________________________
                          /                                                              /
                         /                                            ______________    / /
                        /                                           /   CENTRAL      / /
                       /                                           /   OFFICE       / /
                      /  ###        ### VOICE                     /________________/ /
                     /   ###       ###                                              /
                    /    ###      ###      ###         ###  MAIL                   /
                   /     ###     ###      #####      #####                        /
                  /      ###    ###      ######    ######                        /
                 /       ###   ###      ### ### ### ###                         /
                /        ### ###      ###  #####  ###                          /
               /         ######      ###   ###   ###                          /
              /          #####      ###         ###                          /
             /           ####      ###         ###                          /
            /            ###      ###         ###                          /
           /             ##      ###         ###                          /
          /                                                              /
         /______________________________________________________________/

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Select a softkey >

  Logon                        System       Silence       Channel
                               Status       Alarms        Status
```

Figure A-2xxx
The Message Service Module Logon screen

```
                              Logon/Status                                MM8


                     MMM       MMMERIDIAN
                   MMMMM     MMMMMM
                  MMMMMM   MMMMMMM
                 MMM MMM MMM  MMM     MMM       MMMAIL
                MMM  MMMMMM  MMM    MMMMM     MMMMM
               MMM   MMM    MMM    MMMMMM   MMMMMM
              MMM          MMM    MMM MMM MMM MMM
             MMM          MMM    MMM  MMMMM  MMM
            MMM          MMM    MMM   MMM   MMM
           MMM          MMM    MMM         MMM
          MMM          MMM    MMM         MMM
         MMM          MMM    MMM         MMM
        MMM          MMM    MMM         MMM
       MMM          MMM    MMM         MMM


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------------------------------------------------------------------------
Select a softkey >

   Logon                      System          Silence        DSP Port
                              Status           Alarm          Status
```

Procedure A-1xxx
Logging on

Starting point : Logon Screen.

1 Press [Logon]. Enter the Tools password and press <Return>.

You are prompted to enter the administrator password.

2 Enter the Administrator password and press <Return>.

The Tools screen appears. See the section “The Tools menu” for details.

If an invalid password is entered, an error message appears; try logging on again.

The Tools menu

The Tools menu (Figure A-3) appears after a successful logon. The menu is a routing menu from which you can select the tool you want. Note that the tools listed on the menu depends on what features you have installed on your system. For example, if your system does not have the Synchronize Disks feature, that item will not appear on your Tools menu. Never leave the administrative console unattended while you are logged on.

Note: The actual screen display may differ slightly from the figures shown here.

Figure A-3xxx
The Tools menu

```
                       MM8 Special Tools Package

                           TOOLS Level Access

  1   Redraw
  2   Help
  3   Logoff                     - return to the MMI Logon screen
  4   Release Information        - display the release version number
  5   Move user                  - moves user cabinets one at a time
  6   Modify hardware            - modify hardware database
  7   Set silence compression    - compress out/leave in recorded silence
  8   Control volume             - increase/decrease voice volume
  9   Update MWI                 - update Message Waiting Indicator
 10   Block DMS VoiceMail        - block access to DMS VoiceMail
 11   Find Users                 - find users in the system
 12   Audit all Volumes          - audit all volumes on the system
 13   Synchronize Disks          - disk shadowing maintenance
*14   Other                      - other system/feature dependant options



 Select an item >
```

*   The “Other” option is available if other features are installed.

When you choose the Other option (option <14> in the example above), the menu shown in Figure A-4 is displayed. The items that appear on this menu will depend on the features that you have installed on your system. Figure A-4 shows all possible feature-dependent tools. Your system will most likely only have a few of them and the menu item numbers will therefore be different on your system.

Figure A-4xxx
Feature-dependent tools

```
                  MM8 Special Tools Package
                      TOOLS Level Access
               System/Feature Dependent Tools



 1   Return                       - return to the main TOOLS menu
 2   Change local site ID         - set the site ID to new value
 3   Transfer voice prompts       - read from /write to tape
 4   ACCESS diagnostics           - verify ACCESS link is operational
 5   Configure UATs               - configure User Administration Terminals







 Select an item >
```

```
The actual menu would depend on what additional features are in-
stalled.
```

Procedure A-2xxx
Navigating the Tools menu

Starting point : The Tools menu.

1 Choose an item by entering its number and pressing <Return>.

For selections <1>, <2>, <3> and <4>, the following actions occur:

<1>refreshes the menu screen

<2>presents general information

<3>returns you to the Logon screen

<4>provides a brief summary of any pertinent release information; if the screen is simply redrawn then there is no release information available.

For other selections see the following sections for details:

“Move User”;
“Modify Hardware”;
“Set Silence Compression”;
“Control volume”;
“Update MWI”;
“Block DMS VoiceMail”;
“Find user”;
“Audit all volumes”;

“Synchronize disks”;
“Change local site ID”
“Voice Prompt Transfer Tool”;
“ACCESS Toolkit Diagnostic Tool”
“Configure UATs”

After a few moments, the first screen for the utility you have selected will appear.

2 When you have finished using the utility, terminate the program in the manner described in the appropriate procedure. There are two typical methods of terminating a utility. Depending on the utility, you will either:

a. Use the [Exit] softkey, if the utility displays softkeys or

b. Press <Return> without entering data, or when the utility prompts you to enter <Return>. In some cases, when you toggle to a new setting and press <Return> to confirm the change, the utility will automatically return you to the Tools menu.

Note: *You must terminate one application before starting another.*

3 To log off, enter **3** followed by <Return>.

The Logon screen is redisplayed.

Move User

The Move User utility moves user cabinets, profiles, and voice messages from one user volume to another. This operation is performed one user at a time. Before moving a user, make sure there is enough room on the destination volume.

> ***Note:*** This utility is only useful on systems with more than one user volume.

Figure A-5xxx
Move User utility screen

```
This utility will move a user’s cabinet and its contents from the user’s
current volume to a different user volume.

Before moving a user, make sure there is enough room on the destination
volume.

SYNTAX: MOVEUSER <Customer Number> <Mailbox> <Destination User Volume ID>

EXAMPLE:  John Macmillan’s cabinet is on volume 203. His mailbox is 1234.

        The Destination User Volume ID is 202. He belongs to customer 2.
Enter:    MOVEUSER 2 1234 202

EXAMPLE:  Andy Ng’s cabinet is on volume 204. His mailbox is 12345678.
        His location code is 6338.
        The Destination User Volume ID is 203. He belongs to customer 5.
Enter:    MOVEUSER 5 6338 12345678 203

To EXIT this utility, press RETURN without entering a <Customer Number>.


> MOVEUSER
```

Procedure A-3
Moving users from one volume to another

Starting point : The Tools menu, Move User window activated.

The command line at the bottom of the screen displays the command MOVEUSER and the cursor is positioned immediately after the command. You do not have to enter “moveuser” yourself.

For each user to be moved, enter the user’s customer number and mailbox number followed by the destination user volume ID (see Figure A-5.)

1 Press <Return>.

The user’s cabinet and profile will be created under the “users” directory on the specified volume. This directory must already exist. It will not automatically be created.

If all goes well then the administrator will be notified by the following progress prompts:

Moving Mailbox <mailbox ID> of Customer <customer number> to volume <volume Id>

Mailbox <mailbox ID> of Customer <customer number> moved to volume <volume Id>

The help command provides information on the move user command.

2 Exit the utility by pressing <Return> without entering any data.

Modify Hardware

The function provides facilities for modifying the contents of the hardware database in your DMS VoiceMail system. The hardware database is a system utility which maintains a current listing and description of all nodes, cards, and ports in your system.

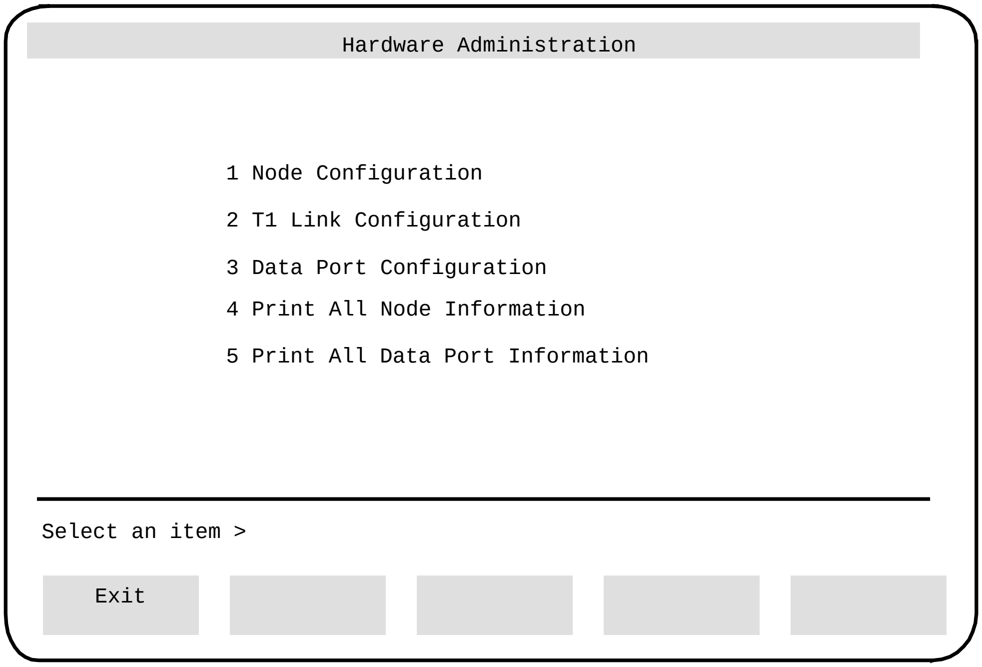
> ***Note:*** For any changes made with this utility to take effect, you must reboot the system after you have made the required changes.

The Hardware Administration menu

The Hardware Administration screen (Figure A-6) provides five functions.

> **CAUTION**
> **Overnight system audits**
> You should not leave the administrative console in any Hardware Administration menu overnight or important system audits may fail due to lack of available memory.

Figure A-6xxx
Hardware Administration menu



Procedure A-4xxx
Navigating the Hardware Administration menu

Starting point : The Tools level utility, Modify Hardware selected.

1 The Hardware Administration screen appears (Figure A-6).

2 Choose an item by entering its number and pressing <Return>.

The menu corresponding to your selection appears. See the following sections for details:

<1> "Node configuration"
<2> "T1 Link configuration"
<3> "Data port configuration"
<4> "Print all node information"
<5> "Print all data port information"

3 Use [Exit] to return to the Tools menu.

Node Configuration

The Node Configuration screen (Figure A-7) is a summary listing of the cards found on all nodes in your system.

> ***Note:*** The figures in this section do not necessarily represent actual hardware configurations. They are illustrations only.

Figure A-7xxx
Node Configuration screen

```
                    Hardware Administration

  Node Configuration


  Node Card_1    Card_2    Card_3

  1    SBC       Bus       Empty
  2    SBC       Bus       Empty
  3    SBC       VP12      VP12
  4    SBC       VP12      VP12
  5    SBC       VP12      VP12
  6    SBC       VP12      VP12
  7    SBC       T1
  8    SBC       T1

Move the cursor to the node number and press the space bar to select.

   Exit        Modify
```

The following abbreviations identify the following cards:

- ***SBC*** - single board computer
- ***Bus*** - high-speed bus
- ***VP12*** - 12-channel voice processor
- ***T1*** - T1 link

Procedure A-5xxx
Modifying node configurations

Starting point : The Hardware Administration screen, <1> entered.

The Node Configuration screen is displayed (Figure A-7).

1 Move the cursor to the node you want to modify and press the <Space Bar>.

Your selection is highlighted.

2 Choose step 2a to modify the configuration information of the node or 2b to return to the Hardware Administration menu.

a. Use [Modify]

The Modify Node screen appears; see the next section, “Modify Node”.

b. Use [Exit].

The Hardware Administration screen is displayed.

Modify Node

The Modify Node screen (Figure A-8) displays the cards and ports (and their attributes) for the node you selected in the Node Configuration screen.

Figure A-8xxx
Modify Node screen

```
                        Hardware Administration

Modify Node

Location   Card_Type    Port_Type Attributes

1-1-*      SBC
1-1-1                   Data:     [Terminal]Printer NWModem MMLink AML/CSL SMDI  PMS AdminPlus
1-1-2                   Data:      Terminal Printer NWModem MMLINK AML/CSL[SMDI] PMS AdminPlus
1-1-3                   Data:      Terminal Printer NWModem MMLINK AML/CSL[SMDI] PMS AdminPlus
1-1-4                   Device:   [Disk] Tape
1-1-5                   Device:   [Disk] Tape

1-2-*      VP12         Voice

1-3-*      VP12         Voice


                                                                     MORE BELOW

    Exit
```

```
                        Hardware Administration          MORE ABOVE

Modify Node

Location   Card_Type    Port_Type Attributes


11-1-*     SBC
11-1-1                   Data       Terminal Printer NWModem MMLink AML/CSL [SMDI] PMS AdminPlus
11-1-2                   Data       Terminal Printer NWModem MMLink AML/CSL [SMDI] PMS AdminPlus
11-1-3                   Data       Terminal Printer NWModem MMLink AML/CSL [SMDI] PMS AdminPlus
11-1-4                   Data       Terminal Printer NWModem MMLink AML/CSL [SMDI] PMS AdminPlus

11-2-*     T1
11-2-1               Link
11-2-2               Link
11-2-3               Link
11-2-4               Link
```

> ***Note:*** The figures in this section do not necessarily represent an actual hardware configuration. They are presented for illustration purposes only.

The screen displays the following information about each card:

- ***Location*** - The physical location of the card in the DMS V oiceMail system. The location is identified by the node-card-port number.

- ***Card Type*** - The function of the card; see Node Configuration for a description of the abbreviations used in this field.
- ***Port Type*** - The type of port. “Data” indicates a serial data communications port. “Device” indicates a mass storage device or tape drive. “Voice” indicates a voice processor port.
- ***Attributes*** *(for ports with port type = Data)*
  - ***Terminal:*** Indicates a connection to an administration terminal or a personal computer.
  - ***MMLink:*** Meridian ACCESS Link. This is the communications channel for Meridian ACCESS. This is an optional feature that is available on CPE systems only.
  - ***AML/CSL:*** Not applicable.
  - ***SMDI:*** Simplified Message Desk Interface. This is the communications channel between DMS VoiceMail and the switch.
  - ***Printer:*** Printer serial connection.
  - ***NWModem:*** Connection to a modem used for networking calls.
  - ***PMS:*** Not applicable.
  - ***AdminPlus:*** Indicates a connection to a personal computer running AdminPlus.
- ***Attributes*** *(for ports with port type = Device)*
  - ***Disk:*** Mass storage subsystem (hard disk)
  - ***Tape:*** Cartridge tape subsystem

Procedure A-6xxx
Modifying nodes

1 Use [Exit].

The Node Configuration screen appears.

T1 Link Configuration

The T1 Link Configuration screen lists the T1 links in the DMS VoiceMail system.

Figure A-9xxx
T1 Link Configuration screen

```
                              Hardware Administration

 T1 Link Configuration


             Primary Connection  Secondary Connection  T1 Clock Refer-
 ence
 T1 Link ID  (Node-Card-Span)    (Node-Card-Span)      Candidacy

      A          13-1-1              14-1-1                Y
      B          13-1-2              14-1-2                Y
      C          13-1-3              14-1-2
      D          13-1-4              14-1-4
      E          15-1-1              16-1-1
      F          15-1-2              16-1-2
      G          15-1-3              16-1-3
      H          15-1-4              16-1-4

________________________________________________________________________
Move the cursor to the item and press the space bar to select.

   Exit      Modify T1 Chnl    Modify T1
             Configuration     Link Setup
```

> ***Note:*** The figures in this section do not necessarily represent actual hardware configurations. They are illustrations only.

The following fields are displayed on this screen:

- ***T1 Link ID* -** A unique identifier for the T1 link. Each link actually consists of two connections, a primary and secondary connection, to provide redundancy.
- ***Primary Connection* -** The location (node-card-span) of the primary connection.
- ***Secondary Connection* -** The location (node-card-span) of the secondary connection.
- ***T1 Clock Reference Candidacy* -** This field shows whether or not the link has been configured as a candidate for clock referencing. Use the [Modify T1 Link Setup] softkey to nominate a link or to disqualify a current candidate. See the section “Modifying the T1 Link Setup” for more information about clock referencing.

Procedure A-7xxx
Modifying T1 Link configurations

Starting point : The Hardware Administration screen, item <2> selected.

The T1 Link Configuration screen appears (Figure A-9).

1 Move the cursor to the link you want to modify and press <Space Bar>.

Your selection is highlighted.

2 To modify the link configuration information, choose step 2a. To modify the T1 link setup information, choose step 2b. To return to the Hardware Administration menu, choose step 2c.

a. Use [Modify T1 Chnl Configuration].

The Modify T1Channel configuration screen appears; see the next section, "Modifying T1 Channels".

b. Use [Modify T1 Link Setup].

The T1 Link Setup screen appears; see the section "Modifying T1 Link Set-up".

c. Use [Exit].

The Hardware Administration screen is redisplayed.

Modifying T1 Channels

The Modify T1 Channel screen (Figure A-10) displays the T1 Channel-Configuration for the link you select.

Figure A-10xxx
Modify T1 Channel screen

```
                       Hardware Administration

Modify T1 Channel-Configuration for Link ID A

Channel   Routing     Login  Logout    Agent    Not-ready
Number    Address     Code    Code     ID Code  Deactivation Code

   1      0 -12345    1234    1234     1234     1234
   2      0 -54321    2222    3333     4444     5555
   3      0 -0
   4      0 -0
   5      0 -0
   6      0 -0
   7      0 -0
   8      0 -0
   9      0 -0
  10      0 -0
  11      0 -0
  12      0 -0
  13      0 -0
  14      0 -0
  15      0 -0
  16      0 -0
  17      0 -0
  18      0 -0
  19      0 -0
  20      0 -0
  21      0 -0
  22      0 -0
  23      0 -0
  24      0 -0

   Save        Cancel
```

Note: The figures in this section do not necessarily represent an actual hardware configuration. They are presented for illustration purposes only.

The following fields are displayed on this screen:

- ***Channel Number -*** The number of the T1 channel.
- ***Routing Address -*** The location of the corresponding agent in the switch. This is the Message Desk Number and is represented in the format xx-yyyy, where xx is the message desk number and yyyy is the terminal number.

- ***Login Code -*** The channel access code for logging in to the UCD group. Leave this field blank (or blank it out if it contains data) if the SMDI_AUTOLOG option is set to “Y” (yes) on the switch. When this field is left blank, DMS VoiceMail inserts a default login code.

  If SMDI_AUTOLOG is set to “N” on the switch, make sure that the code you enter here matches the code configured on the switch. This code can be obtained from your DMS administrator.

- ***Logout Code -*** The channel access code for logging out of the UCD group. Leave this field blank (or blank it out if it contains data) if the SMDI_AUTOLOG option is set to “Y” (yes) on the switch. When this field is left blank, DMS VoiceMail inserts a default login code.

  If SMDI_AUTOLOG is set to “N” on the switch, make sure that the code you enter here matches the code configured on the switch. This code can be obtained from your DMS administrator.

- ***Agent ID Code -*** This code corresponds to the line number (SMDI_LINE_NO) of the agent as configured on the DMS/SL-100. The LINE_NO can either be configured through the servord (so) or through Table IBNFEAT by specifying the SMDI option.
- ***Not-ready Deactivation Code -*** This field is not applicable to DMS UCD environments and should be left blank. It is used in DMS ACD environments for putting the channel to the ACD queue after the channel has logged into the ACD group.

See the *Translations Guide* (NTP 297-7001-310) for more information about these codes.

Modifying the T1 Link Setup

The T1 Link Setup screen (Figure A-11) is used to modify the T1 clock reference candidacy of a T1 link. You may nominate one or more links to serve as the clock reference for the SPM (MSM). An external device in the network (such as the DMS-100, for example) serves as the reference provider.

The actual link that is used as the reference is defined in the T1 Link Status screen (see the “System Status and Maintenance” chapter). If any problems occur on the link that is the current clock reference, or if certain maintenance procedures are being carried out on the link or the card, the system will automatically select one of the other nominated links as the new reference and generate a SEER to indicate that a link has been activated as the reference provider. The following situations will cause the system to select another reference.

- a red alarm is detected
- a yellow alarm is detected
- there is a hardware fault
- the T1 card on which the link resides is disabled
- the TIFN is disabled
- the switch T1 link command is issued
- the T1 link that is the T1 clock reference is disabled

In order to nominate a T1 link for clock reference candidacy, you must first take both the primary and secondary spans associated with the T1 link out-of-service. T1 links are enabled and disabled in the T1 Link Status screen (described in the “System Status and Maintenance” chapter).

Figure A-11xxx
Modify T1 Link Setup screen

```
                         Hardware Administration
 T1 Link Setup for Link ID A

 T1 Clock Reference Candidacy:   [No]   Yes










Select a softkey >

   Save *      Cancel
```

* If you have not disabled the primary and secondary spans, only the [Exit] softkey is displayed and the screen is read-only.

The following field is displayed on this screen:

- ***T1 Clock Reference Candidacy -*** "Y es" indicates that the selected T1 link is nominated as a clock reference candidate. "No" indicates that the link has not been nominated.

Procedure A-8xxx
Nominating/disqualifying a T1 link as a clock reference candidate

Starting point : The Main Menu.

1 Select System Status and Maintenance.

2 Select T1 Link Status.

3 Press [Disable T1].

You are prompted for the number of the link you want to disable.

4 Enter the number of the link you want to disable followed by <Return>.

To disable another link, repeat steps 3 and 4.

5 Press [Exit].

The System Status and Maintenance menu is displayed.

6 Press [Exit].

The Main Menu is displayed.

7 Select Hardware Administration.

8 Select T1 Link Configuration.

9 Move the cursor to the T1 link you want to nominate/disqualify and press <Space Bar> to select it.

Your selection is highlighted.

10 Press [Modify T1 Link Setup].

The T1 Link Setup screen is displayed.

11 Select “Yes” to nominate a link or “No” to disqualify a current candidate.

12 Press [Save].

The select link is nominated/disqualified and the T1 Link Configuration screen is displayed.

13 Return to the T1 Link Status screen in System Status and Maintenance and re-enable the link(s).

14 If necessary, activate one of the candidates as the clock reference using [Change T1 Clocking Mode] in the T1 Link Status screen. See the section “T1 Link Status” in the “System Status and Maintenance” chapter for more information.

Data Port Configuration

The Data Port Configuration screen (Figure A-12) summarizes the data ports on all nodes in your system. For Networking systems, the modem port settings can be modified. Any data port can be selected and its configuration modified. The abbreviations used in this screen are described under “Node Configuration” earlier in this chapter.

Figure A-12xxx
Data Port Configuration screen

```
                    Hardware Administration

Data Port Configuration

Port Location       Description         Device Type       Status

  1-3-1             Node 1  SBC Port 1    Terminal          InService
  1-3-2             Node 1  SBC Port 2    MMLink            InService
  1-3-1             Node 1  SBC Port 3    Terminal          InService
  1-3-2             Node 1  SBC Port 4    Printer           OutofService
  2-3-1             Node 2  SBC Port 1    Printer           InService
  2-3-2             Node 2  SBC Port 2    Printer           OutofService
  2-3-3             Node 2  SBC Port 3    Terminal          InService
  2-3-4             Node 2  SBC Port 4    Terminal          InService
 12-1-2             Node 12 SBC Port 2    NWModem           InService
 13-1-1             Node 13 SBC Port 1    SMDI              InService

Move the cursor to the data port location and press space bar to select.

    Exit          Modify
```

The Data Port Configuration screen displays the following information:

- ***Port Location*** - The port's physical location (node-card-span) in the system.
- ***Description*** - The node and card type on which the port resides.
- ***Device Type*** - The function of the port. SBC port 1 must be set to Terminal. SBC port 1 on node 13 should be set to SMDI.
- ***Status*** - The current operational state of the port.

Procedure A-9xxx
Modifying data ports

Starting point : The Hardware Administration screen, <3> entered.

1 The Data Port Configuration screen appears (Figure A-12).

2 Move the cursor to port to be modified and press <Space Bar>.

Your selection is highlighted.

3 Choose step 3a to modify the configuration information, or 3b to return to the Hardware Administration screen.

a. Use [Modify].

The Modify Data Port screen appears. See the next section for details.

b. Use [Exit].

The Hardware Administration screen appears.

Modify Data Port

The following sections describe the different Data Port screens which can be displayed. The screen that is displayed is determined by the data port that is selected in the Data Port Configuration screen when you press [Modify].

Terminal Data Ports

The Modify Data Port screen for terminals (Figure A-13) allows you to modify information about the terminal connected to the selected port.

Figure A-13xxx
Modify Data Port screen (Terminal)

```
                         Hardware Administration

Modify Data Port

Data Port Location:            1-1-1

Device Type:                   Terminal

Device Name:                   CON181

Baud Rate:                     1200 [2400] 4800 9600

Parity:                        Even Odd  [None]

Number of Windows:             4

Window Width:                  80

Window Height:                 24
______________________________________________________________________

Select a softkey >

   Save          Cancel
```

The following fields are displayed in the screen:

- ***Data Port Location*** - The physical location of the port. A terminal must be located on node 1, SBC port 1. Other terminals can be in the system on other data ports.
- ***Device Type*** - “Terminal” will be displayed.
- ***Device Name*** - The name that identifies the terminal.
- ***Baud Rate*** - This setting depends on the current set-up of the terminal on the port.

- ***Parity*** - The method by which data is communicated. This can be set to "Even", "Odd", or "None", depending on the current set-up of the terminal connected to the port. It is usually set to "None".
- ***Number of Windows*** - This field specifies the number of windows that can be used simultaneously. This will be "6" for the System Administration terminal.
- ***Window Width*** - This field specifies the window width used.
- ***Window Height*** - This field specifies the window height used.

Printer Data Ports

The Modify Data Port screen for printers (Figure A-14) allows you to modify the baud rate and parity of the terminal connected to the selected port.

> ***Note 1:*** A secondary printer can be attached directly to the administration terminal. It does not require a separate data port.

> ***Note 2:*** SEERs and Operational Measurement reports must be directed to a particular printer. The printer is specified in the General Options screen (see the "General Administration" chapter.)

Figure A-14xxx
Modify Data Port screen (Printer)

```
                     Hardware Administration

Modify Data Port

Data Port Location:          1-3-4

Device Type:                 Printer

Device Name:                 PRT134

Baud Rate:                   1200 2400 4800 [9600]

Parity:                      Even Odd [None]

_____________________________________________________

  Save      Cancel
```

The following fields are displayed in the screen:

- ***Data Port Location*** - The physical location of the port.
- ***Device Type*** - The function of the port. This will be set to “Printer”.
- ***Device Name*** - The name of the device.
- ***Baud Rate*** - The setting will depend on the current set-up of the printer that is connected to the port.
- ***Parity*** - The setting will depend on the current set-up of the printer connected to the port.

Procedure 9-10xxx
Modifying the printer data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1. Move the cursor to the printer data port you want to modify.
2. Press the <Space Bar> to select it.
3. Press [Modify].

   The Modify Data Port screen (for the selected printer) is displayed.
4. Use [Save] to save any changes or [Cancel] to disregard any changes.

   The Data Port Configuration screen is displayed.

MMLink Data Port

The Modify Data Port screen for Meridian ACCESS Link (Figure A-15) allows you to modify link characteristics.

Figure A-15xxx
Modify Data Port screen (MMLink)

```
                    Hardware Administration

Modify Data Port

Data Port Location:            1-3-2

Device Type:                   MMLink

Device Name:                   CONSOLE

Baud Rate:                     1200 [2400] 4800 9600

Parity:                        Even Odd [None]

Number of Windows:             1

Window Width:                  80

Window Height:                 24

______________________________________________________________________

  Save          Cancel
```

The following fields are displayed in the screen:

- ***Data Port Location*** - The location of the port in the system.
- ***Device Type*** - The function of the port. It will be set to “MMLink”.
- ***Device Name*** - The name of the device.
- ***Baud Rate*** - Set this field to “9600” for MMLink.
- ***Parity*** - This field is not used for MMLink.
- ***Number of Windows*** - This field specifies the number of windows that can be used simultaneously. This will be “1” for ACCESS.
- ***Window Width*** - This field specifies the window width used.
- ***Window Height*** - This field specifies the window height used.

Procedure 9-11xxx
Modifying the MMLink data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the MMLink data port you want to modify.

2 Press the <Space Bar> to select it.

3 Press [Modify].

The Modify Data Port screen (for the selected MMLink) is displayed.

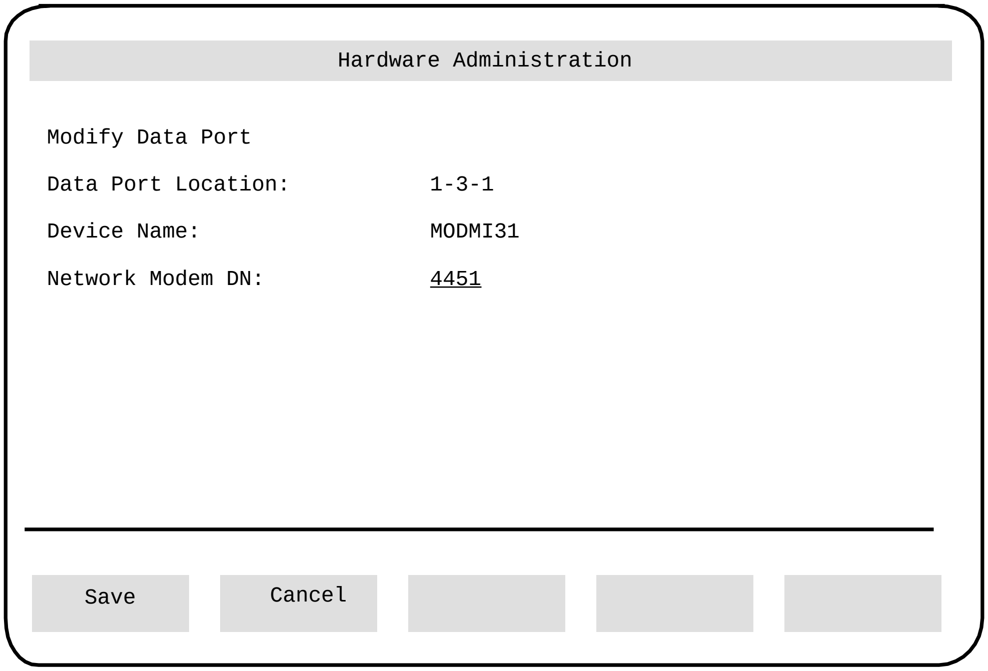
4 Use [Save] to save any changes or [Cancel] to disregard any changes.

The Data Port Configuration screen is displayed.

NWModem Data Port

The Modify Data Port screen for Networking Modems (Figure A-16) allows you to specify the Directory Number (DN) of the modem connected to the selected port.

Figure A-16xxx
Modify Data Port screen (NWModem)



The following read-only fields are displayed on this screen:

- ***Data Port Locatio***n - The physical location of the port in the DMS VoiceMail system.
- ***Device Type*** - The function of the port. This will be “NWModem”.
- ***Network Modem DN*** - The directory number (up to 8 digits) used to identify the modem connected to the port.

Procedure A-12xxx
Modifying the NWModem data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the NWModem data port you want to modify.

2 Press the <Space Bar> to select it.

3 Press [Modify].

The Modify Data Port screen (for the selected NWModem) is displayed.

4 Use [Save] to save any changes or [Cancel] to disregard any changes.

The Data Port Configuration screen is displayed.

SMDI Data Port

The Modify Data Port screen for SMDI (Figure A-17) allows you to modify the baud rate, parity, and transmit mode of the serial connection to the SL-100 or DMS switch at the selected port.

Figure A-17xxx
Modify Data Port screen (SMDI)

```
                    Hardware Administration
 Modify Data Port

 Data Port Location:            1-8-3

 Device Type:                   SMDI

 Device Name:                   SMDI183

 Baud Rate:                     [1200] 2400 4800 9600

 Parity:                        Even Odd  [None]

 Transmit Mode:                 Simplex   [Duplex]

 Link Name:                     SMDI Link Name

 ______________________________________________________________

  Save      Cancel
```

The following fields are displayed in the screen:

- ***Data Port Location*** - The physical location of the port.
- ***Device Type*** - The function of the port. This will be “SMDI”.
- ***Device Name*** - The name of the device. In the above example, set to include the data port location (e.g., SMDI183).
- ***Baud Rate*** - The recommended rate is “2400”.
- ***Parity*** - This will be “Even”.
- ***Transmit Mode*** - This will be “Duplex”.

- ***Link Name -*** The name of the SMDI link. You can enter numeric or alpha characters in this field. It is recommended that you enter a meaningful name (as opposed to a number) so that it is easy to identify the link.

> **CAUTION**
> **Changing the link name**
> Do not change the link name once it has been configured and users have been added to the system. If you change the link name, you will have to change the Message Waiting Link Name in every user profile that refers to that link.

Procedure 9-13xxx
Modifying the SMDI data port

Starting point : The Hardware Administration screen, <3> entered.

The Data Port Configuration screen appears.

1 Move the cursor to the SMDI data port you want to modify.

2 Press the <Space Bar> to select it.

3 Press [Modify].

The Modify Data Port screen (for the selected SMDI link) is displayed.

4 Use [Save] to save any changes or [Cancel] to disregard any changes.

The Data Port Configuration screen is displayed.

Printing node or data port information

The following procedure describes how to print a list of all the node or data port information contained in the hardware database.

Figure A-18xxx
The Hardware Administration menu

```
                    Hardware Administration

               1 Node Configuration

               2 T1 Link Configuration

               3 Data Port Configuration

               4 Print All Node Information

               5 Print All Data Port Information

Select an item >

   Exit      *Cancel              *Continue
             Printing              Printing
```

* The Printing softkeys appear after item 4 or 5 has been selected.

Procedure A-14xxx
Printing node and data port information

Starting point : The Hardware Administration menu, item <3> or <4> selected.

The following softkeys appear: [Continue Printing] and [Cancel Printing].

You are prompted to check that the printer is ready and on-line.

1 Choose step 1a to print the node information or 1b to cancel.

a. Use [Continue Printing].

The node or data port information begins printing.

Once printing is complete, the Hardware Administration menu and its softkeys are redisplayed; you may stop printing at any time by proceeding to 1b.

b. Use [Cancel Printing].

The print operation is cancelled and you are returned to the Hardware Administration menu.

There may be some delay before control is returned to the screen while the system waits for the printer to stop printing.

Set Silence Compression

The Silence Compression Toggle Utility (Figure A-19) allows you to activate or deactivate the Silence Compression feature. This feature removes (compresses) extended periods of silence from messages.

Figure A-19xxx
Silence Compression screen

```
Package MA_PKG loaded.
Component [0]:
**Location code: 1 254 254 254 254 254 CompType: Node State: InService
  hd_PrimeSPNode Node number: 7E000000 BootError: 0
  Test Location: 254 254 254 254
 System type: hd_GP
 Number of hours: 120
 Dsp Pkg. Id: General-32
Component [0]:
**Location code: 2 3 1 254 254 254 CompType: DSP State: InService
  Pkg: General_32 Encoding: mu-law TxLevel: 0 RxLevel: 0
  DTR RejLevel: -51 DTR MaxAccLevel: 1 SBC SilComp: enabled
  SBC AGC: enabled Centre: -15 Span: 16
Package MA_PKG.AREA loaded.

                 Silence Compression Toggle Utility

              Current configuration has silence compression.


Do you wish silence compression to be turned on or off?
ON = Silence will be compressed.  OFF = No compression.
Use up/down arrows to toggle answer.

- > OFF
```

Procedure A-15xxx
Activating/deactivating Silence Compression

Starting point: Tools Menu, <7> entered, Scset window activated.

1 The Silence Compression Toggle Utility screen appears (Figure A-19).

Note: *The actual screen display may differ slightly from the illustration.*

2 Choose the required setting by using the up/down cursor keys. Choose CANCEL if you don’t wish to change the current setting.

Note: *Be sure that the prompt line displays the correct setting before you press <Return>. If silence compression is turned on when you enter this utility, the command line does not display the current setting but displays OFF (the utility assumes you have entered the utility to make a change).*

3 Press <Return>.

The selection is made and the utility is terminated.

Note: *If a change is made, you will have to reboot the system for the change to be effective.*

Control Volume

The Control Volume utility allows the administrator to change the volume levels on both recording and playback voice paths. Each incremental level change, from Level 0 to Level 10, corresponds to an increase of two decibels.

Figure A-20xxx
Control Volume utility screen

```
                          DSP Volume Control


This program will allow the administrator to change the volume level
for all DMS VoiceMail Voice Sessions. The volume level can be set from
0 (lowest volume level) to 10 (maximum volume level). Each incremental
volume level corresponds to an increase of 2 Decibels.


The current volume level is set at    5



**NOTE** If a change is made to the volume level, then a Reboot of the
DMS VoiceMail System is required for the changes to become effective.

 ______________________________________________________________________

 Select a Softkey:

    Exit      Change
              Volume
```

Procedure A-16xxx
Changing the volume level

Starting point: Tools Menu, Volume Control option activated. The current volume level is shown in the center of the screen.

1 To change the volume level, press [Change Volume].

2 Enter the desired volume level and press <Return>.

The screen will be redrawn, showing the new volume level at the center of the screen.

3 Press [Exit] to return to the Tools menu.

Note: *The system must be rebooted for a change in volume level to take effect.*

Update MWI

The Update MWI utility will restore the Message Waiting Indicators (MWIs) for all DMS VoiceMail users. It will turn the MWI on if there are unread messages in a user's cabinet and it will turn the MWI off for users with no unread messages.

This utility should be run after the switch is rebooted (SYSLOAD), since a reboot causes all message waiting indicators to be turned off. It is also useful if the link (SMDI) goes down at a peak time period, because users who were connected to DMS VoiceMail at the time may not have updated MWIs.

Figure A-21xxx
SET MWI Utility screen

```
                         SET MWI Utility



This utility will restore the Message Waiting Indicator for all
DMS VoiceMail users. It will turn ON the MWI if there are unread
messages in the users cabinet and it will turn OFF the MWI for
users with no unread messages.


Do you want to CONTINUE?  Yes
```

Procedure A-17xxx
Restoring message waiting indicators

Starting point: Tools Menu, Update MWI window activated.

1 The screen will display information about the SETMWI utility and will prompt:

Do you want to continue? YES

The following message is displayed:

Running SETMWI Utility Program ....

2 Use the up/down cursor (arrow) keys to respond.

Select YES if you want to reset the MWIs.

Select NO if you do not want to reset the MWIs.

3 Press <Return> to confirm the change.

When the program has completed, the following message is displayed:

Initiated the updating of the Message Waiting Indicators
Press <Return to continue>.

You will then be prompted to press <Return> to continue.

4 Press <Return> to terminate the utility and return to the Tools menu.

Block Access to DMS VoiceMail

The Block Access utility allows the administrator to choose whether to deny all access to DMS VoiceMail voice services in the event of a serious disk failure.

If access is blocked and a disk failure occurs, DMS VoiceMail voice services “shut down” and calls are immediately routed to a live attendant (as configured on the CSE). DMS VoiceMail system administration and maintenance capabilities remain operational.

Figure A-22xxx
Block Access utility screen

```
Enable/disable Access to DMS VoiceMail on Catastrophic Failure



On catastrophic disk failure, access to DMS VoiceMail will NOT be blocked.
Do you want to block access to DMS VoiceMail on disk failure?  YES
```

Procedure A-18xxx
Blocking access to DMS VoiceMail voice services

Starting point: Tools Menu, Block MM window activated.

1 The screen will display the setting currently in effect. Depending on your configuration, one of the following lines will be displayed:

On catastrophic disk failure, access to DMS VoiceMail will be blocked.

On catastrophic disk failure, access to DMS VoiceMail will NOT be blocked.

2 Use the up/down cursor (arrow) keys to display the desired setting.

Select YES if you want to block access.

Select NO if you do not want to block access.

3 Press <Return> to confirm the change.

One of the following messages will appear:

"Upon reboot, access to DMS VoiceMail will be blocked"

"Upon reboot, access to DMS VoiceMail will not be blocked"

One of these messages is displayed for a few seconds after which the Tools menu is automatically redisplayed.

Note: *The system must be rebooted for the change to take effect.*

Find User

This tool allows you to search for and view existing users. You need only complete fields to the extent necessary to identify a user or group of users. Once you have filled in the Find Users screen (Figure A-23), you can either view the found users on the screen or print a list of found users. If you are searching for a particular user in order to modify that user's configuration, select the [View Users] softkey. When you find the user you will have the option to modify that user from the List of Users screen. You do not have to return to the User Administration menu to do so.

This functionality also exists in User Administration. However, two additional fields exist at the Tools level that do not exist in User Administration: Display Data and Only if Primary DN differs from MWI DN.

The fields on the Find Users screen accept the wildcard character + (the plus sign). The plus sign (+) is a pattern-matching character indicating that any characters or digits are acceptable. This character can be used in the following ways. If you want to find all mailbox numbers beginning with "2", enter **2+** in the *Mailbox Number* field. To find all persons named Smith in the Accounting department , enter **Smith** in the *Last Name* field and **Account+** in the *Department* field.

The "_" (underscore) is another wildcard character that you can use when entering search criteria in the Find Users screen. The underscore represents any (single) character. For example, entering Stewar_ in the last name field would find users with the surnames Stewart and Steward.

Figure A-23xxx
The Find Users screen

```
                         User Administration
 Find Users

 User Type: [Any]  Directory_Entry_User  Local_Voice_User Remote_Voice_User

 Status:    [Any]  EnabledDisabled  ExpiredViolation

*If a specific location is desired, include the location code prefix in the
    mailbox number field.

 Mailbox Number:    ______________________  Volume ID: ____

 Last Name:         _____________________________________

 First Name:        ___________

*Department:        _______________

 Extension Number (DN):  ____________________________________

 Personal Verification Status:      [Any]  Not_Recorded   Recorded

 Display Data:        [General]   MWI

 Only if Primary DN differs from MWI DN:    [No] Yes

 Select a softkey >

              Cancel        View Users    Print Users
```

*   This field is displayed only if MMUI (Voice Messaging) is enabled.

The following fields are displayed on this screen:

- ***User Type*** - This field allows you to specify the type of user you want to view. Your choices are “Any”, “Directory Entry User”, “Local Voice User”, or “Remote Voice User”. The default is “Any”.
- ***Status*** - This field is displayed only if *User Type* is set to “Local Voice User”. It allows you to retrieve and view local users according to their mailbox status. You have five choices:
  - *Any* - Select this option if the mailbox status is not a search criterion.
  - *Enabled* - Select this option if you want to find users whose mailboxes are enabled.

- *Disabled* - Select this option to find users whose mailboxes are disabled. These users cannot log on, however messages are still received. A mailbox may be disabled if the user has made too many logon attempts with an incorrect password or if their password has expired.
- *Expired* - Select this option to find users whose passwords have expired. This situation can occur only if users are required to change their password before the number of days stipulated in the field *Maximum Days Permitted Between Password Changes* in the V oice Security Options screen. If this field is set to “0”, users passwords will never expire. If a user’s password has expired, their mailbox will be disabled and they will not be able to log on.
- *Violation* - Select this option to find users who have surpassed the maximum number of allowed invalid logon attempts for their mailbox (configured in the Voice Security Options screen). Users who have made too many invalid logon attempts will not be able to log on and their mailbox will be disabled.

Note: To re-enable a mailbox, you must change its logon status in the View/Modify Local Voice User screen.

- ***Mailbox Number*** **-** This field is displayed if *User Type* is set to “Local Voice User” or “Remote Voice User”. If networking is installed, this field can hold up to 18 characters. If it is not installed, it holds up to 8 characters.
- ***Volume ID*** **-** This field is displayed only if *User Type* is set to “Local Voice User”. It specifies the hard disk volume to which a user is assigned. All users must be assigned to a volume.

  Information on disk usage can be obtained through the Disk Usage report (see “Operational Measurement Reports” in Chapter 11). If you notice that one volume is getting full, you should move some of the users to another volume. Set the *Volume ID* field to the ID of the volume that is almost full in order to get a list of user’s names and their mailbox numbers. You can then move some of these users to another volume with the Move User utility accessible through the Tools menu
- ***Last Name*** - The last name of the user you want to find. To find a group of users with similar last names, use wildcard characters.
- ***First Name*** - The first name of the user you want to find. To find a group of users with similar last names, use wildcard characters.

- ***Department*** - The department to which the user or group of users that you want to find belongs. This field is applicable only if MMUI (Voice Messaging) is enabled.
- ***Extension Number DN*** - The extension number of the user you want to find. (Remember that a user can be associated with up to three extensions.) You can use wildcard characters to find a group of users with extension numbers within a particular range.
- ***Personal Verification Status*** - You may view users according to whether or not they have a personal verification recorded. If you want to ensure that all users have a recorded personal verification, you can generate a list of of users who don't have a recorded verification (by selecting "Not Recorded"). You can then record verifications for these users or contact them and ask them to do this themselves. The default is "Any", meaning that both users with and without a recorded personal verification will be retrieved.
- ***Display Data*** **-** This field allows you to select the format that will be used to display the found users. Your options are:
  - ***General*** **-** The list of found users is displayed in the manner in which they are displayed in User Administration (see Figure 24). This format displays information about the number of days read messages are retained, the storage used and the storage limit, and whether or not a personal verification has been recorded for the user.
  - ***MWI*** **-** The list of found users is displayed as shown in Figure 25. This message waiting indication format displays information about the number of read messages, unread messages, text messages and the MWI status.
- ***Only if Primary DN differs from MWI DN*** **-** When this field is set to "Yes" the system checks for mismatches between users' primary extension DNs and their MWI DNs. For example, users who have more than one extension DN may complain that they are not being notified of new messages. This usually occurs when a DN other than the primary extension is entered as the message waiting indication DN. In such cases, modify the user configuration so that the MWI DN is the primary extension DN.

Figure A-24xxx
General format list of users

User Administration

List of Users

Name	Mailbox	Dept.*	User Type	Retain ReadMsg (days)	Storage Used	Storage Limit (minutes)	Personal Verific. Recorded
Alcott,Tom	2209	Finan	Loc	7	42	12	No
Gordon,John	2145	Sales	Loc	7	20	30	Yes
Jones,Tracy	2134	Admin	Loc	7	7	99	No
Smith, Bod	2291	Accou	Loc	0	22	40	Yes
Valdez, J	212026	Marke	Rem				Yes

Move the cursor to the item and press the spacebar to select it.

Exit	View/Modify User	Delete User	Voice	**Next Page

* The Department column is only displayed if MMUI (Voice Messaging) is enabled.
**Next Page only appears if the information fills more than one screen.

Figure A-25xxx
MWI format list of users

```
                         User Administration
List of Users

Name           User   DN and             Read Unread    Text  MWI
               Type    Mailbox            Msgs  Msgs     Msgs Status

Braun, Maria   Rem     64272408            23   99        10   On
                       64272408
Directory User Dir     3435
Lee, Joan      Loc     *2005                5    2         0  *Off
                       2005
Remote User    Rem     63312332
                       2332
Takemitsu,Toru Loc     2006                12    0         0   Off
                       2006

Move the cursor to the item and press the space bar to select it.

   Exit      View/Modify   Delete User     Voice        Next
                User                                    Page*
```

* This softkey is displayed if the information takes up more than one screen.

Note: The information in this screen shows the status of the user profile when the find user operation was performed. If any voice messaging activity causes a change in the message counts or MWI status, this screen will not be updated while being viewed.

The following information is displayed when found users are listed using the MWI format:

- ***Name*** - The user's last name followed by the first name.
- ***User Type*** - This field displays an abbreviation of the type of user: "Dir" (directory entry user), "Loc" (local voice user), or "Rem" (remote voice user).
- ***DN and Mailbox Number*** - The user's primary extension DN and mailbox number. The mailbox number is not displayed if a directory entry user is retrieved.

  An asterisk (*) is displayed beside the DN when the primary DN is different from the MWI DN.

- ***Read Messages -*** The number of messages in the user's mailbox that have been read.
- ***Unread Messages -*** The number of messages in the user's mailbox that have not yet been read.
- ***Text Messages -*** The number of text messages in the user's mailbox. A text message may indicate a message generated by ACCESS or a hospitality text message.
- ***MWI Status -*** indicates whether message waiting indication is on or off.
  - *On* indicates that there are unread messages or text messages, or both, in the user's mailbox.
  - *Off* indicates that there are no unread messages and no text messages.
  - **Off* indicates that:
    - the MWI Option is set to "Urgent" but there are no unread urgent messages and no text messages, or
    - the MWI Option is set to "None", or
    - the MWI DN is blank.

Audit all volumes

When users delete voice messages, the disk space taken up by those messages isn’t immediately freed up and made available. System audits, which typically begin at 2:30 a.m., make this space available. These early morning audits are sufficient for most normally loaded systems. However, if your system is heavily loaded and there is a lot of traffic, you may have to perform additional audits with this utility. If SEERs with the return code 1103 are being generated, this is an indication that the server is full and that an audit is in order.

Procedure A-19xxx
Auditing all volumes

1 Select “Audit all volumes” from the TOOLs menu and press <Return>.

   Auditing begins immediately. The system does not respond with any prompts because this is a non-destructive procedure.

   Once auditing is complete, the TOOLs menu is re-displayed.

Synchronize disks

Disk shadowing is a mass storage technique in which the same data is duplicated onto a pair of disks in real time. It is used to:

- reduce the chance of data loss and downtime due to disk failure
- double disk read throughput

Disk configurations

All SPMs come with disk shadowing. The configuration on nodes 1 and 2 is shown in Figure A-26. The configuration on nodes 3 to 10 is shown in Figure A-27.

Figure A-26xxx
SPM configuration on nodes 1 and 2

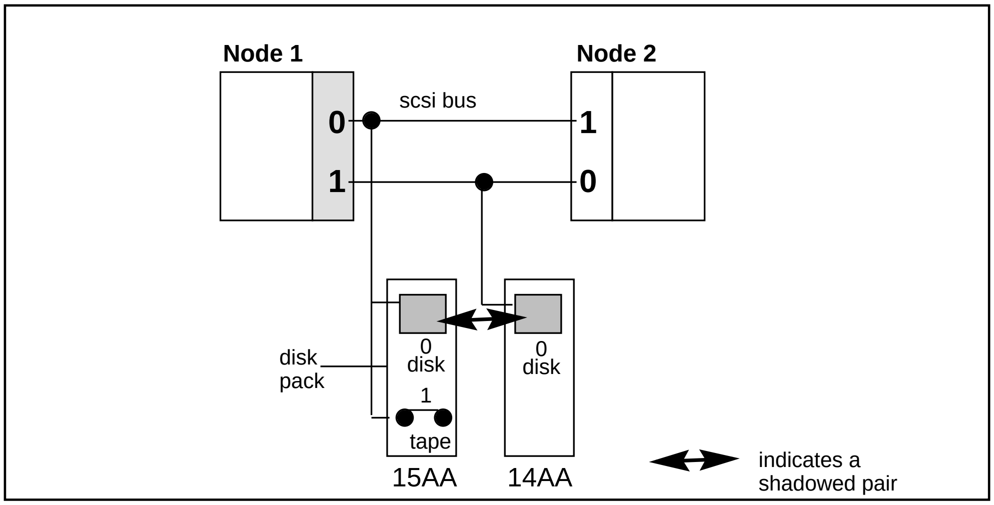
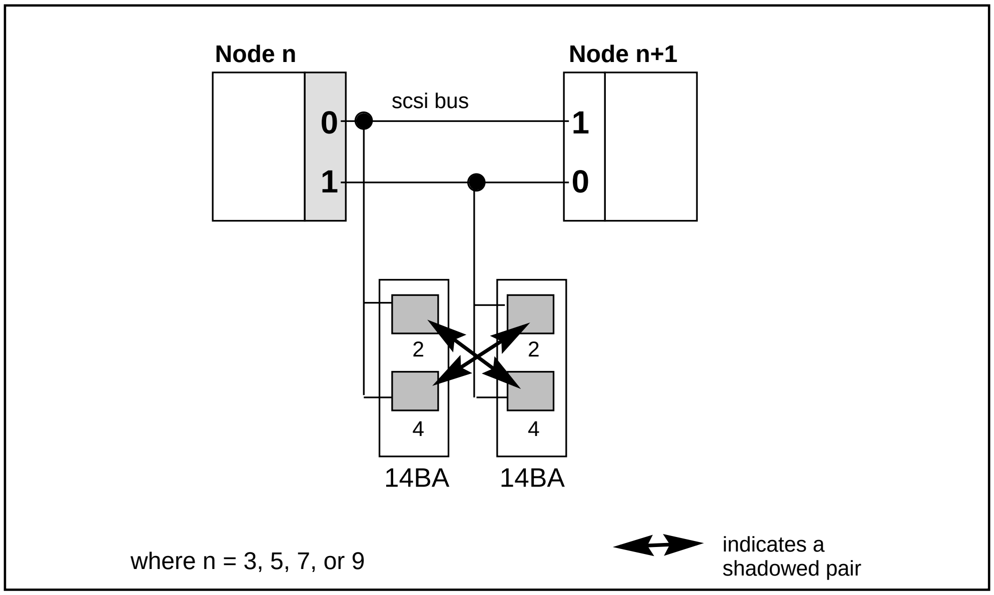


Figure A-27xxx
SPM configuration on nodes 3 to 10



Note the following:

- Each disk pack contributes a disk to each shadowed pair.
- Each node can access its partner's disk drives as a result of SCSI bus coupling, where node "n+1" and node "n" are partners.

The last point gives rise to the following bus and SCSI device numbering scheme:

- Bus 0 on a node is bus 1 on its partner.
- SCSI IDs normally run from 0 to 7. The device number of a device on bus 0 is simply its SCSI ID. The device number of a device on bus 1 is its SCSI ID plus 8.

Commands

All of the following commands print out a return code. 0 indicates normal completion; anything else indicates an error.

enable *src mem*

A shadowed pair is brought online by synchronizing the contents of the two disks that comprise it. This operation is referred to as "syncing" and is started by the enable command.

src specifies the location of the disk pack containing current data. Since there is only one disk pack on a SCSI bus, *src* refers to a bus, and can be either 0 or 1.

The enable command automatically distinguishes between the single-disk AA packs and the dual-disk BA packs. In particular, the enable command will sync both shadowed pairs related to a BA pack.

mem gives the maximum amount of memory that enable is allowed to use. The default value of 64k is recommended.

WARNING
1. Syncing from the wrong source will result in lost data. Also, syncing is usually done from disk packs with enabled disks unless you are trying to clear SEER 6608.

2. Do not set the maximum memory to a value greater than 64k on a live system.

3. Do not sync from both members of a node pair at the same time.

enable -1 *ss dd mem*

This form of the enable command syncs a given disk pair and is invoked by specifying -1 for *src*. *ss* is the device number of the disk to sync from, *dd* is the device number of the disk to sync to, and *mem* is the maximum amount of memory allowed

disable *id*

A disk in a shadowed pair can be taken offline either by the system (automatically, in the event of a failure) or by the disable command (manually). The first method sets off a major alarm, the second does not.

find *id*

id is the number of the device to be checked. If this device is a disk, it will be spun up and its size (in 512-byte blocks) will be printed. This command may be used to verify that a node is able to communicate with all of its disks.

info *node*

This command displays a summary of a node's view of its disks. This information can be used to check if the node's disks are in sync (both disks enabled), and if a node's view of its disks is consistent with its partner's view.

For example, on node 3 of an SPM, one might get the following output:

```
node 3

disk pair 0
boot region:      32-2031
file region:      2032-2936592
disk 2:  RW
disk 12: RW

disk pair 1
boot region:      32-2031
file region:      2032-2936592
disk 10:  --
disk 4:  RW
```

The "disk *n*:" fields in the output indicate each disk's state, and if they are enabled. If both disks are enabled, they are in sync.

The first position of a disk's state is "R" or "-" depending on whether it is handling reads or not. Similarly, the second position is either "W" or "-", depending on whether it is handling writes or not. When the disk's state is "RW", the disk is described as "enabled".

Also, a node can access the same disks as its partner, where node "n+1" and node "n" are partners. Following our example, node 4 accesses the same disks as node 3:

```
node 4

disk pair 0
boot region:      32-2031
file region:      2032-2936592
disk 2:  --
disk 12: RW

disk pair 1
boot region:      32-2031
file region:      2032-2936592
disk 10:  RW
disk 4:  RW
```

Notice from the disk states that disk pair 0 on a node (i.e., node 3) is disk pair 1 on its partner (node 4 in the example above).

init

This command should be used to put the system back into a normal state if a sync operation is interrupted.

node

This command displays the ordinal number of the node that this utility is running on.

Change Local Site ID

This utility can only be used if you have Meridian Networking (only for CPE systems) installed on your system (although the option will show up in the Tools menu even if it is not installed). It allows the administrator to change the local site ID (if, for example, it was entered incorrectly when the site was defined). The site ID you specify must already be defined as a remote site. You will therefore have to create a dummy remote site using Networking Administration (NTP 297-7001-300, *System Administration Guide* or NTP 297-7001-302, *Administration Guide* if you have a single customer system) before using this utility. The current local site will be redefined as a remote site.

Figure A-28xxx
Change Local Site ID screen

```
                    MM8 Special Tools Package
                       TOOLS Level Access
                  System/Feature Dependent Tools



   1   Return                      - return to the main TOOLS menu
   2   Change local site ID        - set the site ID to new value
   3   Transfer voice prompts      - read from /write to tape
   4   ACCESS diagnostics          - verify ACCESS link is operational
   5   Configure UATs              - configure User Administration Terminals




Please enter the Site ID of the remote site you want to become the local
site >
                 Cancel
```

The actual appearance of this menu depends on which optional features are installed.

Procedure A-20
Changing the Local Site ID

Starting point The Tools menu

1 Select “Other” from the Tools menu.

2 Select “Change Local Site ID”.

You are prompted to enter the id of the remote site that will become the new local site.

3 Enter the remote site id and press <Return> or select [Cancel] to exit the utility.

Voice Prompt Transfer Tool

The Voice Prompt Transfer Tool (VPTT) is designed to facilitate the transfer of voice prompt files between DMS VoiceMail systems.

When you select the Transfer Voice Prompt option from the TOOLS Level Access menu, the Transfer Voice Prompts screen (Figure A-29) is displayed.

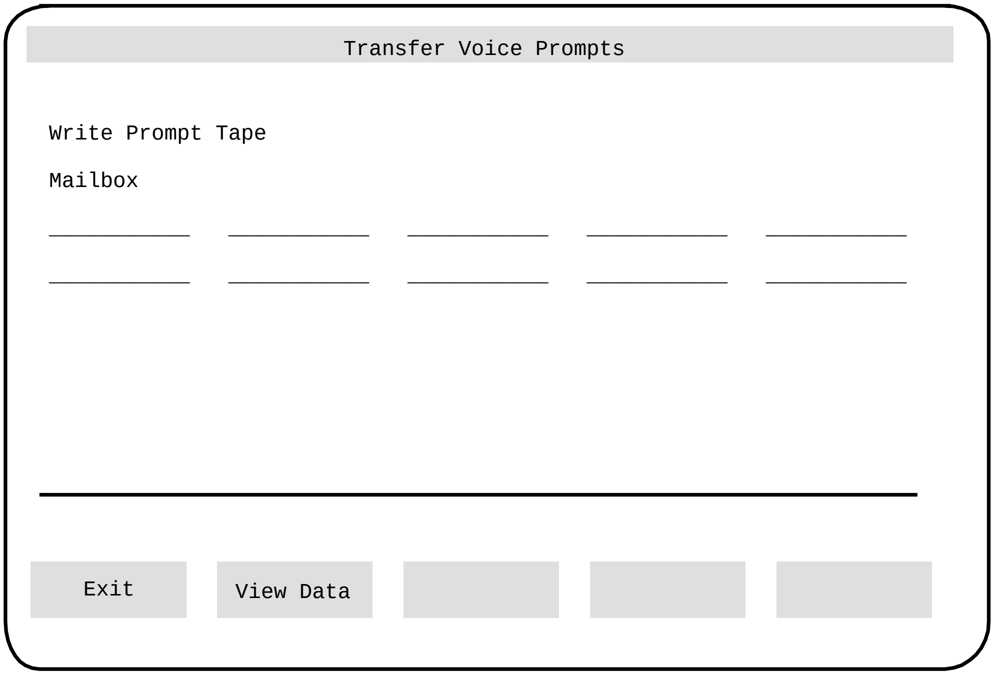
Figure A-29xxx
The Transfer Voice Prompts screen

```
                    Transfer Voice Prompts

      1 Write Prompt Tape

      2 Read from Prompt Tape

      3 Change Default Customer Number

 Select an item >
   Exit
```

Write Prompt Tape

Only voice prompt files can be written to a prompt tape. (Files cannot be appended to the end of an existing tape.) This utility begins its write operation at the beginning of the tape. Figure A-30 shows the Write Prompt Tape screen.

Figure A-30xxx
The Write Prompt Tape screen



You may specify up to 10 different mailbox numbers (identifying the mailboxes containing the voice prompt files) in the available fields. If you want to write the files of more than 10 mailboxes to tape, you will have to perform more than one tape dump operation. You are also limited to writing a maximum of 15 files onto one tape.

Mailbox numbers are validated as you enter them. Only valid input will be accepted. If the entry is invalid, an error message will be displayed. Mailboxes are also validated for the existence of voice prompt files.

> ***Note:*** If Networking is installed on your system, only mailboxes at the local site are considered valid.

Once the mailbox numbers have been entered, the [View Data] softkey can be used to display a screen (Figure A-31) listing the entered mailbox numbers and all the files associated with them.

Figure A-31xxx
The View Data screen

```
                         Transfer Voice Prompts

  View Data

  File Name         File Type         File Size (K)     Mailbox
  Whatkey           Voice Prompt       122               2593
  voice1000         Voice Prompt       181               2593
  temp2             Voice Prompt         1               6401
  myvoice           Voice Prompt      2345               2441

Move the cursor to the file and press space to select.

    Exit                         Write All                  Write Selected
                                   Files                        Files
```

The following fields are displayed on this screen.

- ***File Name*** - The names of the files associated with the mailbox numbers entered in the Write Prompt Tape screen. The total number of files associated with the (up to 10) mailboxes cannot exceed 15.
- ***File Type*** - The type of file associated with the mailbox numbers entered in the Write Prompt Tape screen. Currently, the only valid file type is “Voice Prompt”.
- ***File Size*** - The size of the files (in Kbytes) associated with the mailbox numbers entered in the Write Prompt Tape screen.
- ***Mailbox*** - The mailbox number as entered in the Write Prompt Tape screen.

Procedure A-21xxx
Transferring files to tape

Starting point The Transfer Voice Prompts screen, option <1>, Write Prompt Tape, selected. The Write Prompt Tape screen is displayed.

1 Enter the mailbox numbers of the mailboxes containing the voice prompt files you wish to transfer to tape.

If the total number of voice prompt files contained in the specified mailboxes exceeds 15, you will not be able to enter any more mailbox numbers. In this case, you will have to perform more than one tape dump operation.

2 Press the [View Data] softkey.

The View Data screen is displayed

Note: *A number of error conditions may be reported during the tape dump process such as tape write errors and tape media failures. Error messages are displayed to notify you of such conditions and a [Retry] softkey is displayed so that you may try the tape dump again.*

3 To transfer all the files listed in the View Data screen to tape, go to step 3a. To transfer one or more (but not all files listed) to tape, go to step 3b.

a. Press the [Write All Files] softkey. All of the files listed on the screen are transferred to tape.

The files are converted to the required format. After a short delay, a new screen is displayed, prompting you to insert the tape and press [OK] to start writing to tape.

b. Use the up or down arrow key to move the cursor to the desired file. Press the <Space Bar> to select it. Repeat this step for all files that you want to transfer to tape.

Once all of the files you want to transfer are selected, use the [Write Selected Files] softkey.

The files are converted to the required format. After a short delay, a new screen is displayed, prompting you to insert the tape and press [OK] to start writing to tape.

4 Insert the tape.

5 Press the [OK] softkey to start writing to tape.

The files are transferred to tape. The output from the program that transfers the files is displayed on the screen as the transfer occurs.

Press the [Cancel] softkey to abort an active tape dump at any time during the transfer process.

Note: *The active utility program looks in up to 4 volumes for space to hold the temporary file that is created during the transfer process. This space is released as soon as the tape is made. If there is not enough temporary space available on your system, you will be notified with a message indicating the amount of space required to complete the transfer.*

When the transfer is complete, the following message is displayed:

***** MAKETAPE volume completed *****

6 To make extra copies of the tape, wait until the above message is displayed and then press the [Retry/Another Copy] softkey to make extra copies of the current tape. If you do not need more copies, go to step 7.

7 To transfer more files from the mailboxes already displayed on the View Data screen, go to step 7a. If you want to specify a new group of mailboxes from which you wish to transfer voice prompt files, or if you do not want to do another tape transfer at this point, go to step 7b.

a. Press the [Cancel] softkey.

You are returned to the View Data screen (Figure A-31).

To transfer files to another tape, go to step 3b.

b. Press the [Exit] softkey.

You are returned to the Write Prompt tape screen (Figure A-30). From this screen, you may delete the current mailboxes and enter a different set of mailboxes (go to step 1), or you may return to the Transfer Voice Prompts menu (Figure A-29) by using the [Exit] softkey.

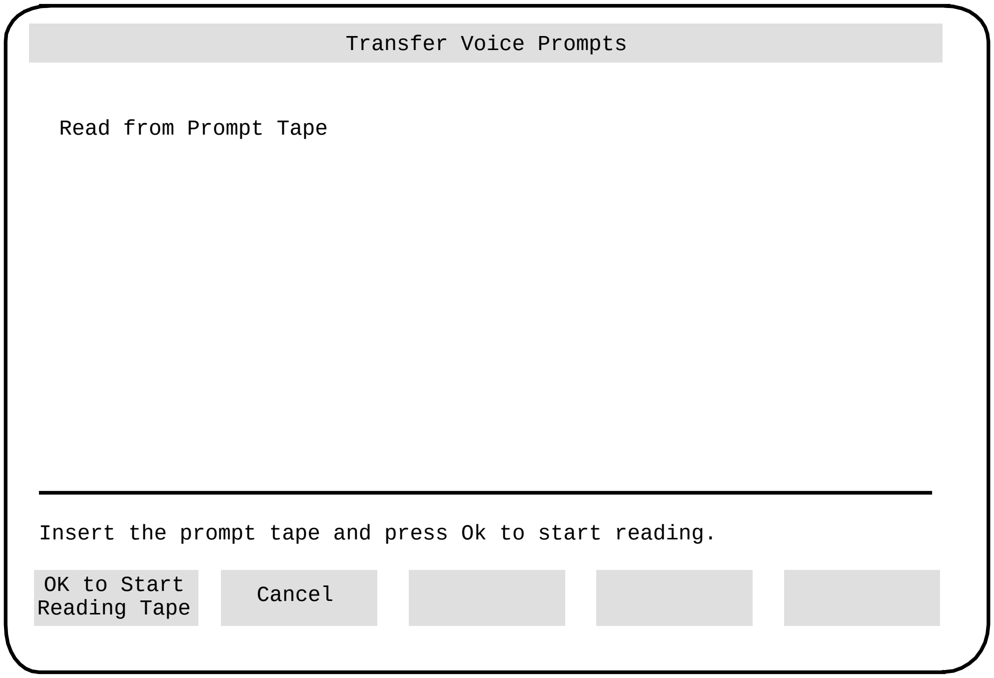
Use the [Exit] softkey on the Transfer Voice Prompts menu screen to return to the TOOLS menu.

Read Prompt Tape

Read Prompt Tape scans all files on the tape and processes them according to your specifications.

When you select the Read Prompt Tape option from the Transfer Voice Prompts menu, the Read from Prompt Tape screen is displayed (Figure A-32).

Figure A-32xxx
The Read from Prompt Tape screen



Procedure A-22xxx
Reading voice prompts from tape

Starting point: The Transfer Voice Prompts menu, <2> (Read from Prompt Tape) selected. The Read from Prompt Tape screen (Figure A-32) appears.

> ***Note:*** *There must be enough memory and temporary space on your system to accommodate the temporary files that are created during this process. If additional memory is required in an* active *system, channels on the MSP node can be courtesied down to get required memory. It is recommended that the higher numbered channels are courtesied down first.*

1 Insert the prompt tape.

2 Press the [OK to Start Reading Tape] softkey.

The tape is read and verified. If an incorrect tape has been inserted, or if there are any errors during the process, you will be notified by a screen message and given the opportunity to retry the operation.

When the correct tape is inserted, the screen shown in Figure A-33 is displayed. This mailbox information is obtained from the tape.

Figure A-33xxx
Source and Destination Mailboxes

```
                         Transfer Voice Prompts

 Read from Prompt Tape

 Source Mailbox                      Destination Mailbox
    2593                               2593
    6401                               6401
    2441                               2441
    6203                               6203

Insert the prompt tape and press OK to start reading.

              Cancel          Install                        Upgrade
```

The Destination Mailbox fields are prefilled with the source mailboxes as the default data. The Destination Mailbox fields can be edited to indicate the mailboxes to which prompts are to be copied. If a Destination Mailbox field is left blank, the contents of the corresponding source mailbox will not be copied to any mailbox.

3 Use [Install] or [Upgrade] to transfer the source mailboxes to the destination mailboxes.

If you use [Upgrade], any existing files with the same name will be overwritten.

If you use [Install], you will be informed that there are existing files having the same name, and they will not be overwritten.

Use [Cancel] to abort all action.

You are returned to the Transfer Voice Prompts menu.

Change default customer number

When selected, you are prompted to enter the customer number to be used when referencing mailboxes in the read or write commands. No validation is performed on the number you enter.

Access Toolkit Diagnostic Tool

This utility can be used to diagnose and/or monitor system activity related to Meridian ACCESS running on a UNIX processor.

The diagnostic tool includes a group of commands which allow you to verify:

- if the ACCESS link is operational
- link stability
- the ACCESS port number and link speed
- the number of link outages that have occurred
- if application traffic is present
- whether the Meridian ACCESS tasks are running
- whether the applications processor link handler is running

Figure A-34 shows the initial screen that is displayed when the diagnostic tool is loaded from the TOOLS menu. The last line on the screen displays the current command.

ACCESS components

There are three primary components on each side of the ACCESS link. They are briefly discussed in the following sections. If you require a more detailed description, refer to the Overview in the *Meridian 1 ACCESS Configuration Guide* (NTP 555-7001-315).

DMS VoiceMail Components

Toolkit (TK)

There is a TK for each voice port on the system. The TK is responsible for executing API commands received across the Meridian ACCESS link.

Toolkit Master (TKM)

TKM acts as a resource manager for Toolkit tasks.

Toolkit Communications (TC)

The TC task is responsible for driving the Meridian ACCESS link. It implements a proprietary protocol that supports variable size packets, checksum error handling, virtual channels and retransmission on errors. Valid command packets received are passed on to the appropriate toolkit task.

Application Processor Components

ACCESS Link Handler

This task provides functionality equivalent to the TC for the applications processor side. The link handler is split into two tasks, one receives data and the other handles the output.

ACCESS API Library

This is the ACCESS object code library containing ACCESS API functions that are linked in with the applications. Most functions translate into commands that are put into a data packet and passed on to the Link Handler.

Application

This is the ‘C’ program written by either Northern Telecom or a VAR which uses ACCESS API functions to answer calls when they arrive. The application controls the Interactive Voice Response (IVR) service being provided.

Figure A-34xxx
The Meridian ACCESS Diagnostics screen

Meridian ACCESS Diagnostics

To select a different command use the up or down cursor key.

To execute a command press the <return> key.

To EXIT this program select the EXIT command.

Diagnostic Command: MMverify

Procedure A-23xxx
Running Meridian ACCESS Diagnostics

Starting point The Tools menu, Meridian ACCESS Diagnostics window activated.

1 Select the appropriate command by using the up or down cursor key (the current command is displayed on the last text line).

2 Press <Return> to execute the command.

See the following sections, "MMVERIFY" and/or "APVERIFY" for a description of the command you plan to execute.

3 When you have finished running diagnostics, select the EXIT command (by using the up or down cursor key until EXIT is displayed on the Diagnostic Command line). Press <Return> to return to the Tools menu.

Commands

Two commands, MMVERIFY and APVERIFY, comprise the diagnostic tool.

MMVERIFY

This command performs the necessary checks and displays a report on the status of ACCESS software running on the DMS VoiceMail side of the link.

MMVERIFY reports the following items.

- ***Link status*** - the report indicates whether or not the ACCESS link is operational.
- ***TC status*** - This is an indication of whether or not the TC task is running. When the TC task is running, the link is either in operational mode or attempting to synchronize with the UNIX processor. If the link is operational, then the link handler on the UNIX processor is up and running.
- ***TKM status*** - This is an indication of whether or not the TKM task is running.

  Note: For the TC to be running, the TKM must be present.

- ***Link speed and configuration*** - The link speed and data port settings are displayed.
- ***Link OM information*** - Operational Measurement (OM) data for the TC is displayed (as shown in Figure A-35).
- ***Link stability*** - There are several indicators in the OM data which can help to determine link stability. Of interest is the number of errors detected. There are several types of errors that occur. For each type, a total is calculated. These totals are then used to calculate the link error rate. It is quite normal to have some errors. The error rate will be slightly higher for more heavily-used links.

  If the error rate remains greater than 0.01 percent, action should be taken. On a system that has been up and running normally, the error rate should not fluctuate greatly. However, during installation or configuration changes you may experience a higher error rate for several reasons:

- The ACCESS RS-232 cable is too long (e.g., greater than 50 feet).
- The application processor cannot cope with link traffic. This is probably the case if the majority of received errors are “Naks”.
- Application traffic needs to be reduced. This is probably the case if the majority of errors are on the receiving (DMS VoiceMail) side.

The MMVERIFY command should always be run first. If it reports that the ACCESS link is not operational, then the APVERIFY command will only confirm this. APVERIFY should be run after MMVERIFY to confirm that the ACCESS software is running on the applications processor. If the link is unplugged, it may take up to 30 seconds for this to be detected.

Figure A-35 shows the output for the MMVERIFY command when the system is operating normally.

Figure A-35xxx
MMVERIFY output screen

```
Information for Link #1

TC last started 03/05/90 09:11:28  TKM last start 03/05/90 09:11:10
Active Sessions=12   AOIC Pending=0
ACCESS port=2, baud rate=9600

TC crashes=0, Link outages=0

PKT COUNTS     Data    Poll     Ack      Nak  Sync    Term
Sent           330745 0        967225   0    0       0
Received       890402 76823    332258   3    1       0

PKT ERRORS     Format Checksum Sequence Error Percentage    Timeouts
Received       0       0        0        0.00               0


ACCESS link is operational on Meridian Mail
```

The following are descriptions of the fields appearing on this screen.

- ***Data*** - displays the total number of data packets.
- ***Poll*** - the number of sanity poll packets (sent only when the link is idle).
- ***Ack*** - the number of acknowledgement packets.
- ***Nak*** - the number of negative acknowledgement packets.
- ***Synch*** - the number of synchronization request packets.
- ***Term*** - the number of shutdown link request packets.

- ***Format*** - the number of packets received in the wrong format.
- ***Checksum*** - the number of packets received containing checksum errors.
- ***Sequence*** - the number of packets received out of sequence.
- ***Error Percentage*** - the link receive error rate, calculated by dividing the total number of packets received by the number of packet transmission errors.
- ***Timeouts*** **-** UNIX workstation responses not received.

Figure A-36 shows an example of MMVERIFY output when DMS Voice-Mail is attempting to synchronize with the UNIX side which is not responding. It will continually attempt to synchronize with the UNIX processor until the UNIX link handler responds. If the link handler is running, the synchronization process only takes a couple of seconds. If it is not running, or if the UNIX processor is not running, this condition will continue to persist.

Figure A-36xxx
MMVERIFY output screen

```
Information for Link #1

TC last started 03/05/90 09:11:28  TKM last start 03/05/90 09:11:10
Active Sessions=12   AOIC Pending=0
ACCESS port=2, baud rate=9600

TC crashes=0, Link outages=0

PKT COUNTS     Data    Poll     Ack      Nak  Sync    Term
Sent           330745  76111    967225   0    33      0
Received       890402  76823    332258   3    1       0

PKT ERRORS     Format  Checksum Sequence Error Percentage   Timeouts
Received       0       0        0        0.00               0



ACCESS link is attempting to synchronize with applications processor.
```

If you repeat the MMVERIFY command while this condition exists, you will notice that the number of Synch packets sent continues to increase.

APVERIFY

This command performs the necessary checks and displays a report on the status of ACCESS software running on the applications processor.

When APVERIFY is running, it monitors the link and reports if any application traffic was detected. If the link appears operational but no link traffic is detected within 30 seconds, the link handler on the applications processor is not functioning correctly.

Figure A-37 shows the output for the APVERIFY command when the system is operating normally and one or more applications are active.

Figure A-37xxx
APVERIFY output screen

```
Monitoring ACCESS link activity
Traffic test may take up to 30 seconds. Please standby.
....................................
Application traffic is present on link.
Application processor link handler is operational.
```

Running APVERIFY when MMVERIFY indicates that the link is not operational simply confirms this, as shown in Figure A-38.

Figure A-38xxx
APVERIFY output screen (link not operational)

```
ACCESS link is not operational.
Use MMVERIFY command for further details.
```

LINKS

This command is used to display the ACCESS links that are currently running on the system. A brief status is shown for each link.

Figure A-39xxx
LINKS output screen

```
LINK    Node    TKM     TC       Status         Sessions   Pending

1        1      RunningRunning  Synchronized  32         0

2        3      RunningRunning  Synchronized  32         0
```

OMRESET

This command is used to reset (to zero) the numbers displayed by the MMVERIFY command.

SETLINK

This command is for systems on which multiple ACCESS links are configured. To find out the number of links configured, use the LINKS command. This command is used to set the context for the MMVERIFY, APVERIFY and OMRESET commands.

Diagnosing ACCESS configuration problems

If results indicate that there may be a configuration problem on DMS Voice-Mail, it is useful to know the actual configuration requirements of ACCESS on DMS VoiceMail. The following sections discuss configuration parameters which can be checked.

Enabling ACCESS

To check if ACCESS is enabled on your DMS VoiceMail system, select the General System Administration option from the DMS VoiceMail Main Menu. From the next menu displayed, select General System Options. “Meridian ACCESS” will be listed under the *Available Features* portion of the screen.

ACCESS link cable

The ACCESS link cable should be connected to the dataports configured as MMLINK in DMS VoiceMail.

Viewing hardware database settings

To view hardware database settings, select the Hardware Administration option from the DMS VoiceMail Main Menu. From the next menu displayed, select the Data Port Configuration option. This screen displays a list of configured system data ports only, one of which should be of device type “MMLINK”. To view the port setting, select the item in the list and press the [View] softkey.

The data port that is configured for ACCESS must have the following settings:

- *Port Type* must be set to “Data”
- *Data Type* must be set to “MMLink”
- *Baud Rate* must be set to “9600” (this baud rate is only used for synchronizing the link and will automatically adjust itself to the correct default once the link is up).

Configure UATs

Note: This tool is available if the Multiple Administration Terminals feature is installed on your system.

This utility allows you to view or change the number of User Administration Terminals (UATs). The utility lists the currently configured UATs and provides a means for adding the UAT program to a terminal or deleting it from one. Terminals are normally defined as UAT terminals during installation. Therefore, this utility is only used in the event that you need to change the configuration that was created during installation.

The Configure UATs option is displayed when you select “Other” from the main TOOLS menu. When you select Configure UATs, the screen shown in Figure A-40 is displayed.

Figure A-40xxx
The User Administration Terminal (UAT) Configuration screen

```
              User Administration Terminal Configuration

A User Administration Terminal Program is configured on the following
Terminals:

  UAT301

The following Terminals can be configured
with a User Administration Program:

  CON0123    CON0431



**NOTE** If a change to the User Administration Terminals is made then a
Reboot of the system is required for the changes to become effective.

Select a softkey >

   Exit        Add UAT      Delete UAT
```

A terminal name is displayed in this screen for any data port that is defined as “Terminal” in the hardware database. The first part of the screen displays all terminals that have been configured with the User Administration Terminal Program. The bottom portion of the screen displays those terminals that can be configured with the program. The following procedures describe how to add and delete the UAT program.

Procedure A-24xxx
Adding a UAT

Note: *If there are no available terminal ports, an existing unused dataport must be configured using the hw_modify tool. No more than three User Administration Terminals can be installed on a system.*

1 Press the [Add UAT] softkey.

You are prompted to specify the name of the terminal that you want to add.

A new softkey, [Cancel], is displayed. If you do not wish to proceed, use [Cancel] to quit the operation.

2 Enter name of one of the terminals that can be configured with a User Administration Program. Press <Return>.

You are prompted to provide a suffix for the new terminal name. All terminals configured with the User Administration Terminal program begin with “UAT”.

3 Enter the suffix for the new terminal name (you do not have to enter “UAT”).

The terminal name is added to the top of the screen where the configured terminals are listed.

4 Press [Exit] to return to the main TOOLS menu.

5 Reboot the system for the changes to take effect.

Procedure A-25xxx
Deleting a UAT

1 Press the [Delete UAT] softkey.

You are prompted to specify the name of the terminal that you want to delete.

A new softkey, [Cancel], is displayed. If you do not wish to proceed, use [Cancel] to quit the operation.

2 Enter name of one of the terminals that is currently configured with a User Administration Program. Press <Return>.

The terminal name is removed from the top of the screen and moved to the list of terminals that can be configured with the User Administration Program. The name is changed from UATxxx to CONxxx.

3 Press [Exit] to return to the main TOOLS menu.

4 Reboot the system for the changes to take effect.

Appendix B: **Examples of Voice Menu Applications**

Introduction

The DMS VoiceMail Voice Menu Applications Administration function lets you create specialized call-processing and information-related applications. The two categories of application are automated attendant and Information Menu. Some typical application scenarios are outlined in the following pages.

Definition types

Voice menu applications can be created using the following types of definitions:

Announcement definitions

An announcement is a recording which provides information to callers who dial a predetermined DN. A caller who is connected to an announcement cannot perform any actions, like pressing a key. The caller simply listens to the announcement and hangs up.

Thru-dial definitions

A thru-dial service allows callers to dial someone who has an extension on switch (by dialing the extension or the person’s name, if name dialing is enabled), or to place a local call or even a long-distance call. However, thru-dialers which allow local or long-distance calls should only be made available selectively in order to guard your system against toll fraud.

Time of Day Control definitions

A time-of-day controller allows you to control the activation of a voice menu application based on the date and time of day. In a time of day control definition you specify which services should be played during business hours, off-hours and holidays.

Voice menu definitions

A voice menu offers a maximum of 12 actions to callers, one for each key on the telephone keypad. When a caller connects to a voice menu, he or she is asked to touch 1 for one action, touch 2 for another action and so on. Any action within a voice menu can activate another voice menu, thus creating layer upon layer of information. You can include up to 20 layers of voice menus in a single application.

Automated attendants

An automated attendant answers more than one call at a time (the maximum number of calls is based on the number of ports your system has). Nowhere in DMS VoiceMail do you actually create an automated attendant definition. Instead, you build an automated attendant by using the four definition types described above. An automated attendant can be as simple as an announcement. Or, an automated attendant can use the thru-dial service to allow callers to dial the extension of the person they want to talk to. A voice menu on the other hand, would present callers with a series of choices.

Voice services profile

The Voice Services Profile screen is also a part of building voice menu applications. This profile contains parameters that are general to all voice menu applications. For example, in this profile you determine timeout values (how long the system should wait for a response from the caller before taking action), and the maximum length of recorded prompts that are used in voice menu applications.

Voice prompt maintenance and remote activation

Two other services are used to support your voice menu applications: Voice Prompt Maintenance and Remote Activation. Both allow you to remotely access the system through a telephone set.

- ***Voice prompt maintenance*** - allows you to modify/manage the various prompts, greetings and announcements within voice menu applications from any touch-tone telephone.
- ***Remote activation*** - allows you to remotely enable, disable or change the voice menu application associated with a particular DN.

A maximum of 1000 voice menu applications can be created on any one DMS VoiceMail system, each with a unique Directory Number (DN) which callers dial to gain access to the application.

DMS VoiceMail systems are engineered to have specific amounts of disk space dedicated to voice menus. The storage time available will depend on the size (number of nodes) your system has. Read Chapter 4, “Technical Specifications” in the *General Description* (NTP 555-7001-100), for more information.

DMS VoiceMail systems are engineered to have specific amounts of disk space dedicated to voice menus. The storage time available will depend on the size (number of nodes) your system has. Read Chapter 4, “Technical Specifications” in the *General Description* (NTP 555-7001-100), for more information.

Examples of Automated Attendants

An automated attendant answers calls within a predefined number of rings. A simple automated attendant is typically a thru-dialer which plays a greeting and allows a caller to dial the extension (or name) of the person he or she wants to speak to. More sophisticated automated attendants are typically voice menus which allow callers to choose from a variety of actions. Automated attendants improve communication by:

- answering multiple calls at the same time;
- answering calls 24 hours a day;
- allowing callers to control their own calls without depending on a human attendant.

Automated attendants improve your organization's efficiency by:

- removing a significant percentage of the workload from clerical staff;
- allowing employees to be reached after normal business hours.

After reviewing several types of automated attendants outlined below, you will be able to determine your organization's requirements, then create automated attendants that best suit its needs.

Choosing the type of automated attendant

When an organization chooses and configures an automated attendant, certain questions must be considered:

1 Are there areas of the organization that get a large number of calls?

If one person or department gets a high volume of calls, then the directory number of the person or department should probably be presented as an option on a voice menu.

2 Will most callers know the extension number or name of the person they need to contact?

If most callers don't know the extension number or name, then a live attendant rather than an automated attendant would be better. Perhaps in this case a separate line could be established with an automated attendant, and this number could be given to regular callers. Perhaps an automated attendant would be better if limited to after-hours use.

3 Will most of the callers have touch-tone phones?

Automated attendants are only useful for callers with touch-tone phones. The automatic revert to the Revert DN on the initial timeout routes rotary-phone callers to a live attendant, so the rotary-phone caller will reach the person wanted. If most callers have rotary-dial phones, perhaps only an after-hours automated attendant is desired.

4 Are there people in the organization that remain in the office after hours?

Whether there will be people available after hours or not determines the type of after-hours automated attendant. If no one is available to take calls, then an announcement is sufficient. If there are people in the office after hours, then a basic automated attendant using the thru-dial service would be appropriate.

5 Are there different requirements for business hours versus after hours?

Organizations may have different requirements for automated attendants based on the time of day, day of the week, and day of the year. Different types of automated attendants can handle these requirements. The time-of-day controls establish the schedule for these attendants.

General setup requirements

The automated attendants described in this document have the same installation requirements, as follows:

1 The automated attendant directory number must be listed in the Voice Service-DN Table. This entry associates the service ID to the number dialed, allowing external callers to reach an automated attendant.

2 Restricted dialing prefixes, if any, must be specified in the thru-dialer screen associated with the automated attendant (for example, long distance calls, ESN calls, or local calls).

3 A password defined for automated attendants requiring restricted access. This password should be given only to the people who are allowed access.

Basic automated attendant

The most basic automated attendant uses only the thru-dial service. When a caller reaches the basic automated attendant, a prompt for the extension number or name is played. The caller then enters the number or name and the call is placed. If the caller does nothing, the call is routed to a live attendant.

This version of the automated attendant is useful for organizations whose employees have the same likelihood of being called. The value of this automated attendant depends on callers knowing the extension numbers or names of the people they want to reach.

This automated attendant can be set up as a separate outside line so that this number would be given to callers who know the extension numbers or names of the people they wish to reach. Another line with a live attendant could then handle the one-time callers who do not have the necessary information.

Example

The following automated attendant is very simple to set up. A thru-dialer is defined with a recorded greeting and a specified Revert DN.

Setup

1 Greeting

"Thank you for calling Peer Enterprises. If you would like to speak to the receptionist, or if you have a rotary-dial phone, please wait on the line and someone will be with you shortly. If you know the extension number of the person you wish to dial, enter the extension number followed by the number sign. If you know only the person's name, please press 11, then spell the last name followed by the first, using the letters on the keypad."

2 Revert DN

This DN should be the extension number of someone who can assist callers and route them to the desired party. Usually the number of a receptionist or a secretary is chosen as the Revert DN.

Figure B-1 on the following page is a flowchart for the above example. Figure B-2 shows the Thru-Dial definition.

Figure B-1xxx
Basic automated attendant

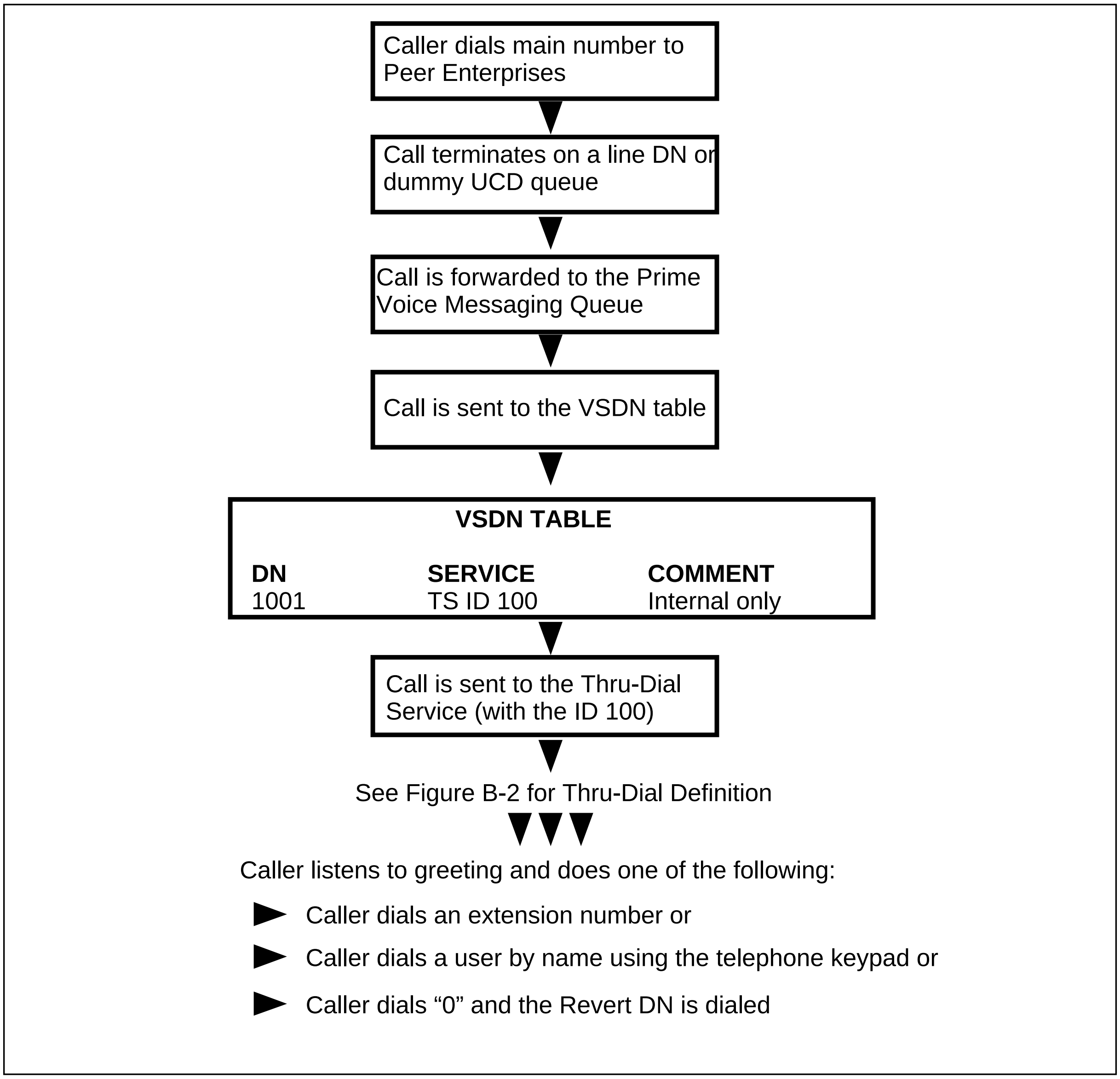


Figure B-2xxx
The Thru-Dial Definition for a basic automated attendant

```
VS Config/Menu Applications Admin

Add a Thru-Dial Definition

Thru-Dial ID:      100                Title: Internal Thru-Dialer

Revert DN:         0

Access Password:   2645321            Update Password:43209876

Greeting Recorded (Voice):    Yes

Language for Prompts:         [AmericanEnglish]
                              CanadianFrench
                              AmericanSpanish

Dial by:                      Number  Name  [Both]

Restriction/Permission Set:   [Custom]  OnSwitch  Local  LongDistance1
                              LongDistance2

Restriction Codes:  9__ ___ ___ ____ ____ ____ ____ ____ ____ ____

Permission Codes:   ___ ___ ____ ____ ____ ____ ____ ____ ____ ____

Select a softkey >
  Save     Cancel              Voice
```

In this definition, callers are allowed to dial the user’s extension or name. Notice that the Custom Restriction/Permission set is used and that all calls external to the switch are restricted. This thru-dialer only allows calls to internal extensions.

Automated attendant with menu choices

This version of an automated attendant can be configured to let callers choose to be routed to predefined numbers rather than entering an extension number or name.

This type of automated attendant is useful for organizations that have people or departments receiving a high volume of calls. The directory number of such people or departments can be put in a menu so that, when a caller selects that menu choice, the call is routed directly to the appropriate number.

Having predefined numbers available to callers is also useful when callers may not know the extension number or name of the person they need to reach.

Example

The following example suits an organization that has many calls for sales representatives and for product servicing. Not only will the automated attendant allow callers to dial an extension or name themselves, but it will also have predefined menu choices that route callers automatically to the representative they want.

Setup

A Voice Menu must be set up with a greeting that informs the caller of menu choices.

1 Greeting

"Welcome to Nadir International. If you would like to talk to one of our sales representatives, press 1. If you would like to talk to someone in the service department, press 2. If you know the extension number or name of the person you would like to reach, press 3. If you need assistance, press 0 or just wait on the line."

2 Menu Choices Prompt

"To talk to a sales representative, press 1. To talk to a customer service representative, press 2. To dial the extension number or name of the person you would like to reach, press 3. If you need assistance, press 0."

3 Key 1

Key 1 is set up to call the sales representative. The action is Call (CL) and the number is the extension number of the sales representative.

4 Key 2

Key 2 is set up to call the customer service representative. The action is Call (CL) and the number is the extension number of the customer service representative.

5 Key 3

Key 3 is set to go to a thru-dial Service. The action is thru-dial Service (TS), and the number is the ID of the thru-dialer. This allows callers to dial the extension numbers or names of the people they want to reach.

6 Revert DN

The Revert DN must be filled in with the number of a person who can assist the caller in reaching the appropriate person. As with the Basic automated attendant, this would normally be a receptionist or secretary.

Figure B-3 shows a flowchart for the above example. Figure B-4 shows the Voice Menu Definition.

Figure B-3xxx
Automated attendant with menu choices

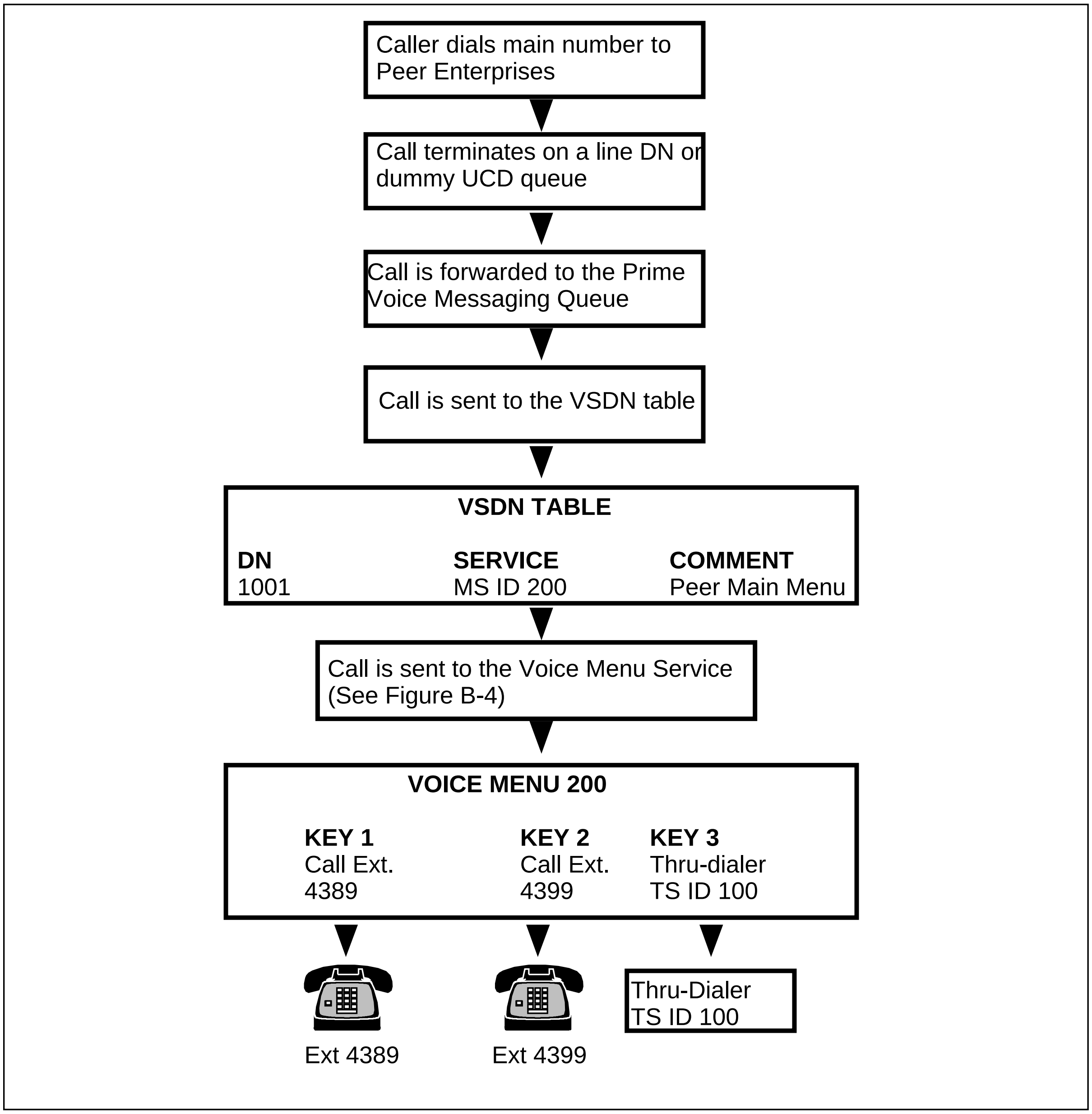


Figure B-4xxx
The Voice Menu Definition for an automated attendant with menu choices

```
                    VS Config/Menu Applications Admin

  Add a Voice Menu Definition

* Choice of Menu Actions:
  AS Announcement Service    CL   Call                  RV   Call Revert DN
  DS Disconnect              EM   Express Messaging     GS   Greetings Service
  PP Play Prompt             RP   Repeat Menu Choices   MM   Return to Main Menu
  TS Thru-Dial Service       TD   Time-of-Day Control   TR   Transcription Service
  VF Voice Forms Service     MS   Voice Menu Service    VM   Voice Messaging

  Voice Menu ID:    200                  Title:   Peer Main Menu

  Revert DN:        0

  Access Password: 3295341        Update Password: 39243221

  Greeting Recorded (Voice):        Yes     Menu Choices Recorded (Voice): Yes

**Silent Disconnect:                [No]    Yes

  Language for Prompts: [AmericanEnglish]
                         CanadianFrench
                         AmericanSpanish
                         French

         Key            Action                              Comments
         1              CL   Calling Number:  4389
         2              CL   Calling Number:  4399
         3              TS   Thru Dial ID:     100
         4              PP
         5              PP
         6              PP
         7              PP
         8              PP
         9              PP
    Initial No Response RP
    Delayed  Response   RV

 Select a softkey >
     Save         Cancel                       Voice
```

* Some of these actions are feature-dependent and may not appear on your screen.

**This field is displayed if multiple languages are installed on your system.

Announcement-only automated attendant

The announcement-only automated attendant plays a recorded voice to the caller. The caller's only option is to listen or hang up.

This type of automated attendant is used by organizations that shut down completely after hours so that there is no one there to take calls. The announcement informs callers that the premises are closed, states business hours, then disconnects.

Setup

1 Greeting

"Welcome to Nadir International. Our office is closed right now. Please call back during our regular business hours, Monday to Friday, 8:30 a.m. to 5:00 p.m."

2 Revert DN

Enter the Revert DN in the Announcement screen (for example, the number of the night security guard).

Figure B-5 shows a flowchart for the above example. Figure B-6 shows the Announcement Definition.

Figure B-5xxx
Automated attendant with announcement only

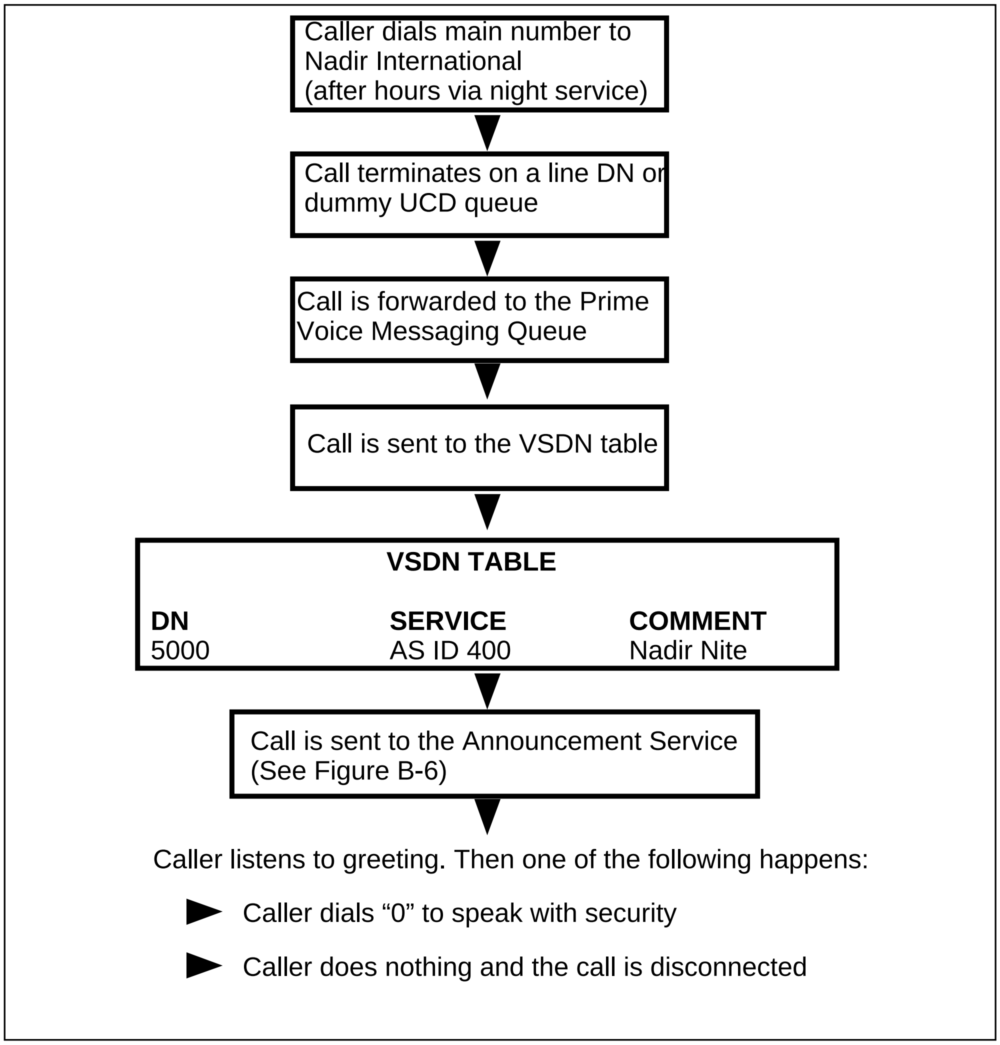


Figure B-6xxx
The Announcement Definition for an announcement-only automated attendant

```
          VS Config/Menu Applications Admin

Add an Announcement Definition

Announcement ID:    400______________________Title: Night announcement______

Revert DN:          0__________________________

Access Password:    3499876_______      Update Password: 78230076_______

Announcement  Recorded (Voice):    Yes

Language for Prompts:              [AmericanEnglish]
                                   CanadianFrench
                                   AmericanSpanish
                                   French

Select a softkey >
  Save       Cancel                   Voice
```

Leave a message

This type of automated attendant is similar to the announcement-only type, except that it allows callers to leave messages for persons they are trying to reach. Like announcement-only, it is used as an after-hours attendant.

Example

A slight modification to the automated attendant with menu options (above) makes the automated attendant more useful to callers. This modification allows a caller to leave a voice message and specify who the message is for.

Setup

A Voice Menu definition must be set up with a greeting and menu choices that allow a caller to leave a message in either a predefined mailbox or the mailbox of the person the caller is trying to reach.

1 Greeting

"Welcome to Nadir International. Our office is closed right now. Our business hours are Monday to Friday, 8:30 a.m. to 5:00 p.m. If you would like to leave a message for someone whose extension or name you know, press 1. If you prefer, you can leave a message that will be forwarded to the appropriate person. To leave a message press 2, then state the name of the person for whom you wish to leave a message, your name, number, and your message after the tone."

2 Menu Choices Prompt

"Our business hours are Monday to Friday, 8:30 a.m. to 5:00 p.m. If you would like to leave a message for someone whose extension or name you know, press 1. If you prefer, you can leave a message that will be forwarded to the appropriate person. To leave a message press 2, then state your name, number, and your message after the tone."

3 Key 1

Express Messaging (EM) is the action associated with Key 1. The mailbox number should be left blank so that callers can enter the ones they want.

4 Key 2

Express Messaging (EM) is the action associated with Key 2. In this case, a mailbox number must be entered. The owner of the specified mailbox should log in every morning, listen to new messages, then forward them to the appropriate people.

5 Revert DN

Enter the Revert DN in the Voice Menu definition screen.

Figure B-7 shows a flowchart for the above example. Figure B-8 shows the Voice Menu Definition.

Figure B-7xxx
A “leave a message” style automated attendant

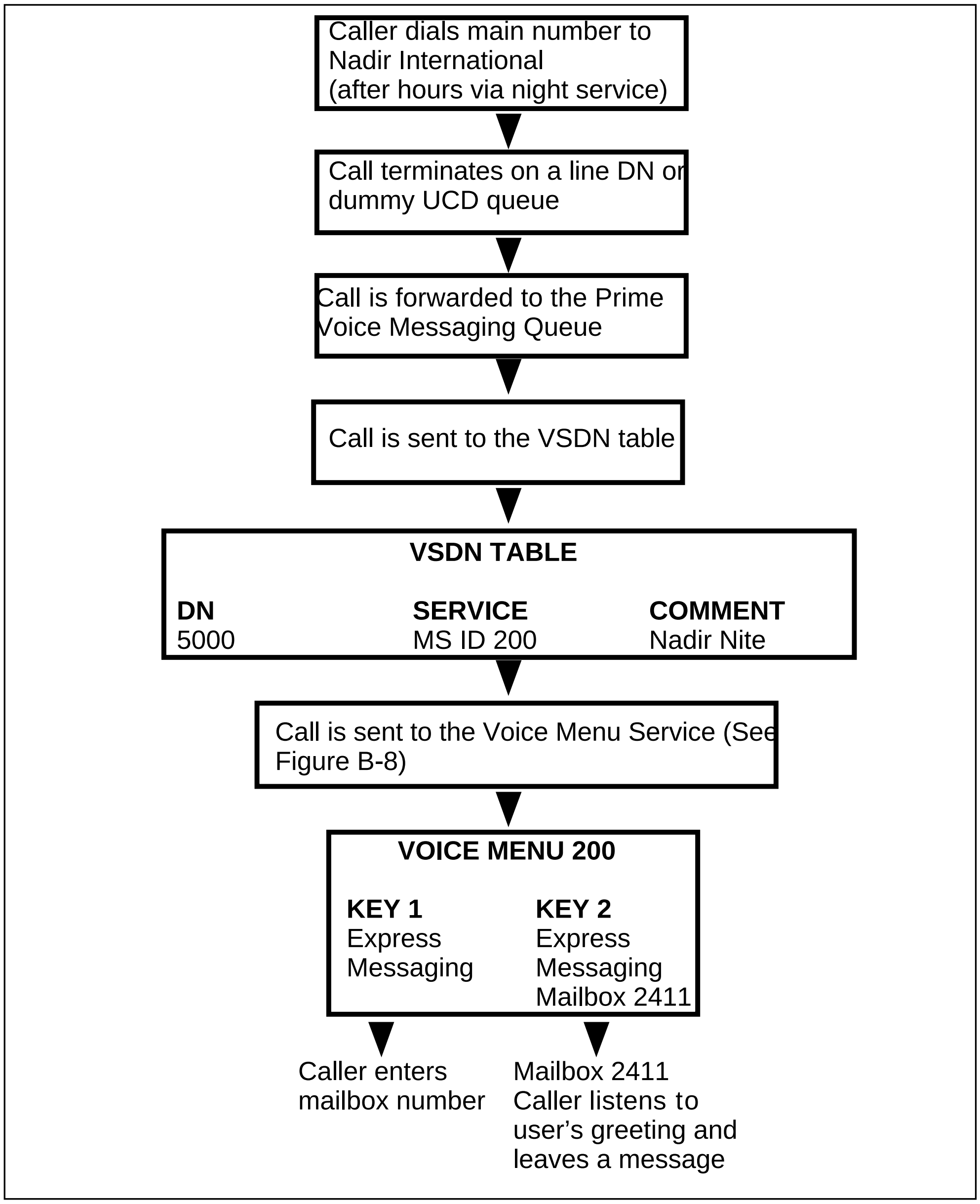


Figure B-8xxx
The Voice Menu Definition for a "leave a message" style automated attendant

```
                    VS Config/Menu Applications Admin

 Add a Voice Menu Definition

* Choice of Menu Actions:
 AS Announcement Service    CL   Call                  RV   Call Revert DN
 DS Disconnect              EM   Express Messaging     GS   Greetings Service
 PP Play Prompt             RP   Repeat Menu Choices   MM   Return to Main Menu
 TS Thru-Dial Service       TD   Time-of-Day Control   TR   Transcription Service
 VF Voice Forms Service     MS   Voice Menu Service    VM   Voice Messaging

 Voice Menu ID:    200                     Title:   Nadir Nite

 Revert DN:        0

 Access Password: 3239320          Update Password: 992430087

 Greeting Recorded (Voice):          Yes     Menu Choices Recorded (Voice): Yes

**Silent Disconnect:                 [No]    Yes

 Language for Prompts: [AmericanEnglish]
                        CanadianFrench
                        AmericanSpanish
                        French

        Key            Action                              Comments
        1              EM   Mailbox ID:      _____
        2              EM   Mailbox ID:      2411
        3              PP
        4              PP
        5              PP
        6              PP
        7              PP
        8              PP
        9              PP
   Initial No Response RP
   Delayed  Response   RV

 Select a softkey >

    Save         Cancel                       Voice
```

*   Some of these actions are feature-dependent and may not appear on your screen.

**This field is displayed if multiple languages are installed on your system.

Time-of-Day Controls

An organization may decide that different automated attendants are required at different times of the day. For example, an automated attendant offering menu choices during the day would be appropriate, but at night when there is no one in the office an announcement-only attendant would be better. This situation can be handled through time-of-day controls.

Example

This example combines the automated attendants previously described under Basic automated attendant and automated attendant with menu choices. The attendant that allows thru-dial Service is offered during business hours, and the announcement-only type is used after hours.

Setup

A time-of-day control must be defined, specifying the different automated attendants for different times of the day. Time-specific automated attendant IDs include:

1. Business-Hours Service ID

   Basic automated attendant service ID.

2. Off-Hours Service ID

   Announcement-only service ID.

3. Holiday Service ID

   Announcement-only service ID. A special announcement is given on the holidays specified in the Voice Services Profile.

4. Business Hours

   The organization’s business hours should be filled in. During these hours the basic automated attendant is used; outside these hours, the announcement-only automated attendant is used.

Figure B-9 shows a flowchart for the above example. Figure B-10 shows the Time-of-Day Control Definition.

Figure B-9xxx
Automated attendant accessed through a time-of-day control

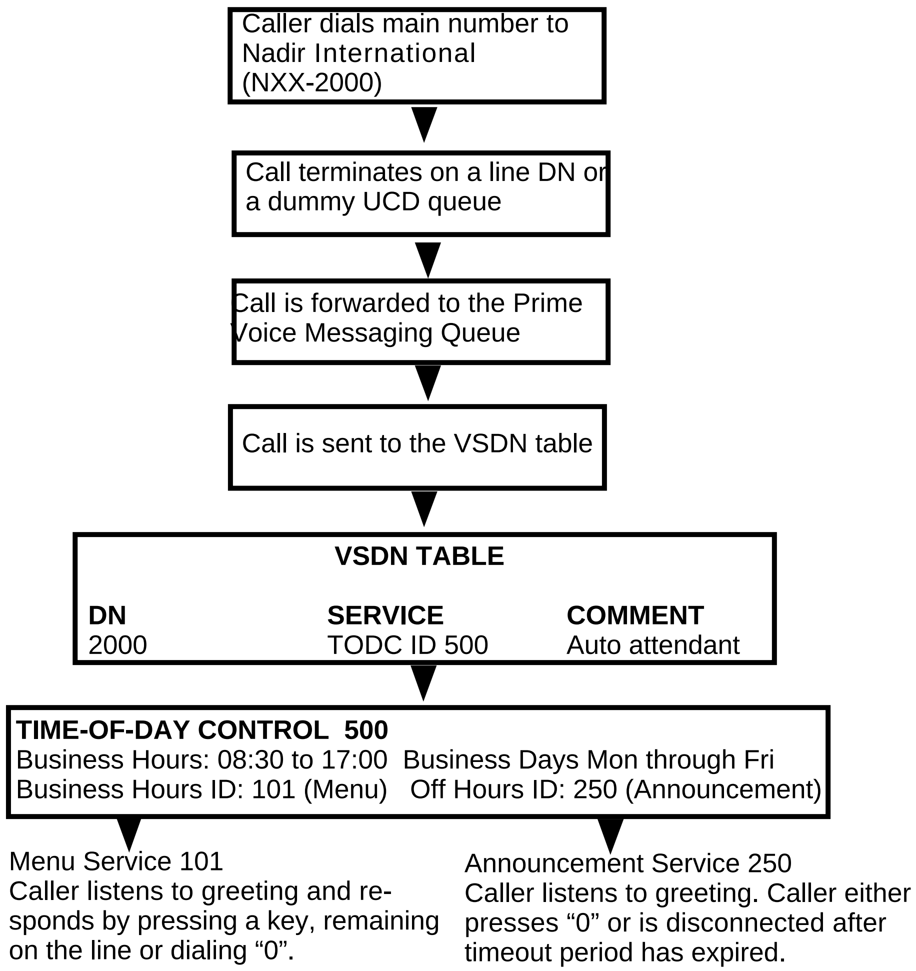


Figure B-10xxx
The Time-of-Day Control Definition for an automated attendant

```
          VS Config/Menu Applications Admin

Add a Time-of-Day Control Definition

Time-of-Day Control ID:  500            Business Hours Service ID: 101

Off-Hours Service ID: 250               Holiday Service ID:      600

     Business Days            Business Hours
     Sunday      [No] Yes
     Monday      No   [Yes]  08:30  to 17:00
     Tuesday     No   [Yes]  08:30  to 17:00
     Wednesday   No   [Yes]  08:30  to 17:00
     Thursday    No   [Yes]  08:30  to 17:00
     Friday      No   [Yes]  08:30  to 17:00
     Saturday    [No] Yes

Select a softkey >
   Save        Cancel                    Voice
```

Nested Time-of-Day Controls

If you have a single facility (for example, a single DID queue) serving multiple departments with different business hours, you can nest several time-of-day controllers in order to meet the needs of all departments.

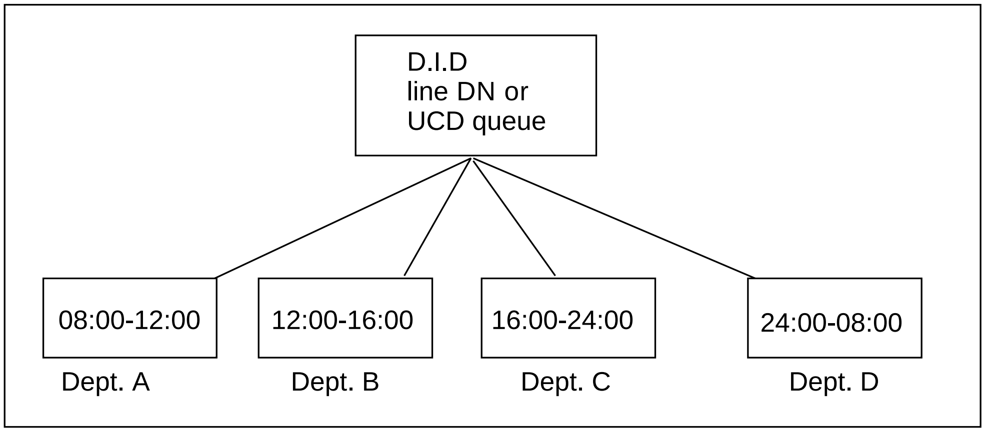
Example

Your company has four departments with different business hours and they are all served by a single DID queue. Department A's business hours are between 8:00 a.m. and 12:00 noon. At noon, Department A wants calls to be routed to Department B. Department B closes at 4:00 p.m. at which time they want their calls to be routed to Department C and so on. See Figure B-11.

To nest time-of-day controls, the following rules apply:

- The time-of-day controllers must cover a 24-hour period.
- The time-of-day controllers must cover seven days a week.
- The maximum allowable nesting levels is six.

Figure B-11xxx
One D.I.D. queue serving four departments



Setup

You must program a separate time-of-day control for each department. Y ou will therefore, end up with four different time-of-day controls.

For each time-of-day control, configure the following:

1 Business-Hours Service ID

Menu service ID for the department.

2 Off-Hours Service ID

Time of Day Control ID of the following department.

3 Holiday Service ID

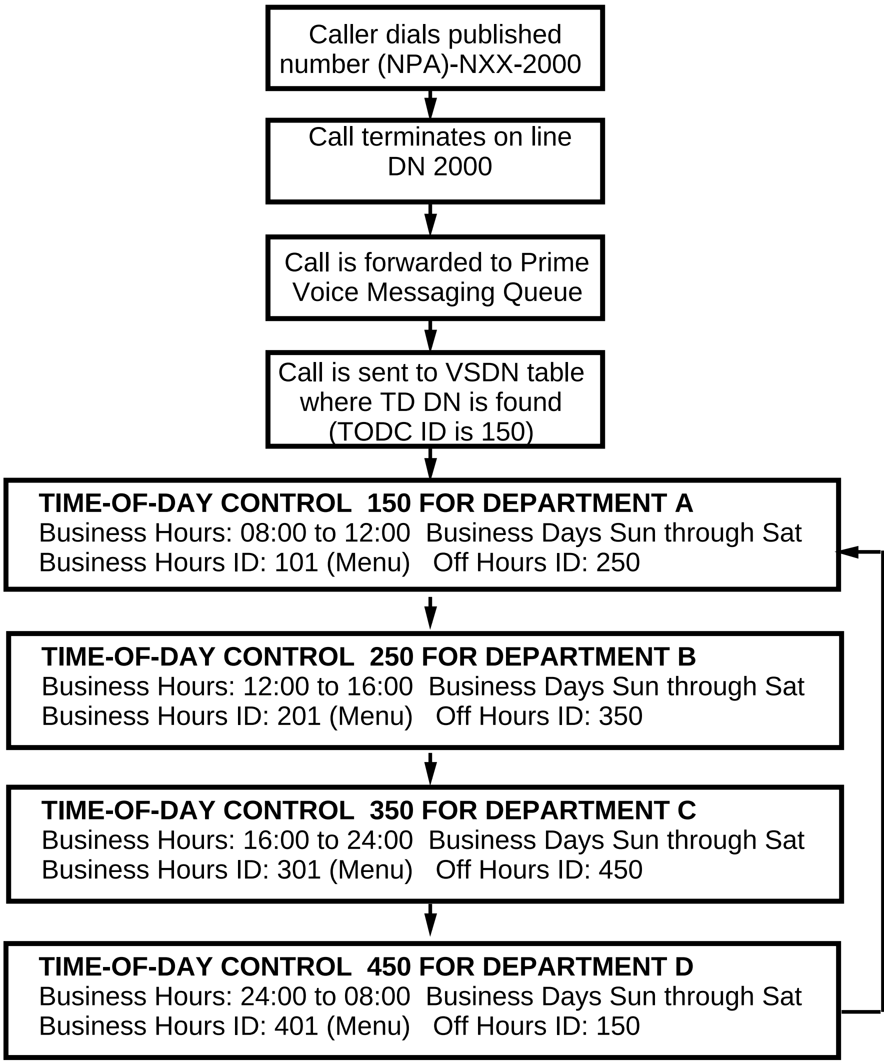
Announcement-only service ID. A special announcement is given on the holidays specified in the Voice Services Profile.

4 Business Hours

The department’s business hours should be filled in. During these hours the automated attendant with menu choices is used. Outside these hours, calls are directed to the time-of-day control of the department whose business hours begin when the business hours of this department end.

Figure B-12 is a flowchart of the applications required to meet the needs of the four departments shown in Figure B-11. See Figure B-10 for an example of a Time-of-Day Control definition.

Figure B-12xxx
Flowchart for nested time-of-day controllers



Mixed live and automated attendants

When an organization has a receptionist, an automated attendant is useful for handling overflow calls and for handling calls when the receptionist is unavailable (after hours, lunch breaks, and so on).

Example

This example has a receptionist available to answer calls except for certain periods. When the receptionist is unavailable, the basic time-of-day controls specified in the previous example are used. If the receptionist is away from the phone, the Basic automated attendant is used, and after hours the announcement-only automated attendant is used.

Setup

Use the setup from the previous example to establish the automated attendant. To establish the connection between the receptionist and the automated attendants, the attendant console must be set up to have the time-of-day controls number as the night number. When the receptionist is dealing with a call and the line is busy, all calls go to the appropriate automated attendant.

Figure B-13 shows a flowchart for the above example.

Figure B-13xxx
Automated attendant with live attendants

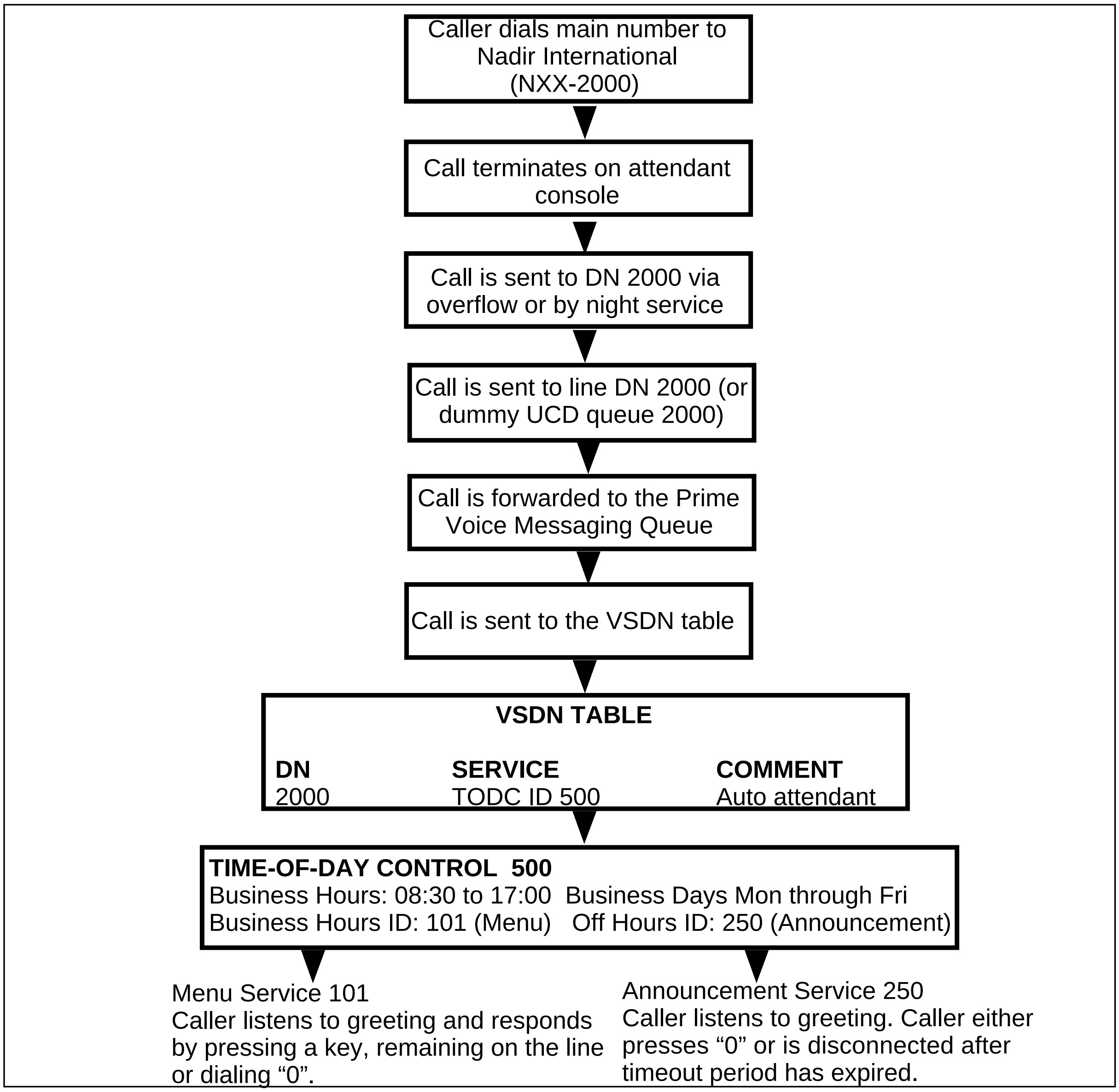


Figure B-10 depicts a Time-of-Day Control Definition that meets the requirements of this example. See Figure B-4 for an example of Voice Menu Definition and Figure B-6 for an example of an Announcement Definition.

Information menus

Information menus offer callers single-digit access to pre-recorded information.

Setup

The applications outlined in this section have common installation requirements:

1 An entry in the Voice Service-DN Table if the application is to be dialed directly. This entry connects the dialed number to the service’s definition number.

2 The Access Password field in the definition screen must be filled in if access to the application is restricted. This password should be given only to the people who are allowed access.

Announcement-only

The simplest form of information application is the announcement. It provides a single recording of information to what might be a large number of people who want or need to hear it. Examples of information provided as announcements are daily stock quotes, weather reports, the time, train and bus schedules, equipment status, fares, store hours, daily restaurant menus, and daily or weekly specials in stores.

Example

At Peer Enterprises almost everyone is interested in the status of the printers and the computer systems. This information must be current to be useful, so it may need to be updated many times a day.

Setup

An announcement must be defined with an Update Password so that the recording of the announcement can be delegated.

Two announcement definitions may have to be set up if there are different Revert DNs for different times of day or days of the week. A time-of-day control definition would then have to be set up (see the section on time-of-day controls earlier in this chapter).

1 Announcement

   "You have reached the status line for the printers and controllers. Friday at 4:40 p.m., all printers and controllers are up, except printer three which is down. A service call has been placed. If you are experiencing any other problems and need assistance, press zero."

2 Revert DN

   The Revert DN should be the number of the person responsible for the printers and controllers.

3 Update Password

   An Update Password should be assigned so that the person responsible for updating the status can re-record it (by using the Voice Prompt Maintenance Service).

4 The Access Password field in the definition form must be filled in if access to the application is restricted. This password should be given only to the people who are allowed access.

5 Add a DN to the Voice Service-DN Table if the announcement is to be dialed directly. In the Add DN Information screen, you will be prompted to specify the announcement ID when you specify the service.

See Figure B-5 which shows a flowchart for an automated attendant with an announcement. This scenario is very similar to the situation depicted in Figure B-5. The differences are that instead of an external caller, the announcement is accessed by an internal user who dials the DN. The Revert DN would be that of the person responsible for printers (not 0). You can also refer to Figure B-6 which shows an announcement definition.

Multi-layer announcements

Multi-layer announcements are applications that offer so much information that it is more practical to split the information into amounts that can be managed easily. This type of application is created using voice menus that allow callers to choose what they want to hear, rather than having to listen to lengthy recordings.

Example

A manager, frequently away traveling, has a busy schedule. A secretary or assistant can record the items of a menu that lists all the manager’s meetings and appointments for each day of the week. The manager can then call in to this menu and hear up-to-date information and details concerning the schedule.

Setup

A Voice Menu must be set up.

1 Greeting

“For Monday’s meetings and appointments, press 1. For Tuesday’s, press 2. For Wednesday’s, press 3. For Thursday’s, press 4. For Friday’s, press 5. To leave me a message, press 6. To log into your mailbox, press 7.”

2 Menu Choices Prompt

“For Monday’s meetings and appointments, press 1. For Tuesday’s, press 2. For Wednesday’s, press 3. For Thursday’s, press 4. For Friday’s, press 5. To leave me a message, press 6. To log into your mailbox, press 7.”

3 Keys 1 through 5

Keys 1, 2, 3, 4, and 5, which represent the days of the week, have the Play Prompt (PP) action associated with them. The prompt recorded contains a list of appointments for that day. These five prompts should have a format similar to: “For Monday, March 11, you have an appointment with the Customer Relations group at 11:00 a.m. in the manager’s office. Don’t forget to call Frank Winchester to discuss the report he wrote for you.”

4 Key 6

Key 6 has the Express Messaging (EM) action associated with it. The mailbox number is that of the secretary or assistant who sets up the menu.

5 Key 7

Key 7 has the Voice Messaging (VM) action associated with it. Thus the manager can log into his or her mailbox to retrieve any messages.

6 Revert DN

The Revert DN is the number of the secretary or assistant.

7 Access Password

An Access Password is assigned so that only the manager has access to this menu.

8 Update Password

An Update Password is assigned so that the secretary or assistant can update the prompts from any phone.

Figure B-14 shows a flowchart for the above example. Figure B-15 shows the corresponding voice menu definition.

Figure B-14xxx
Multi-layer announcements

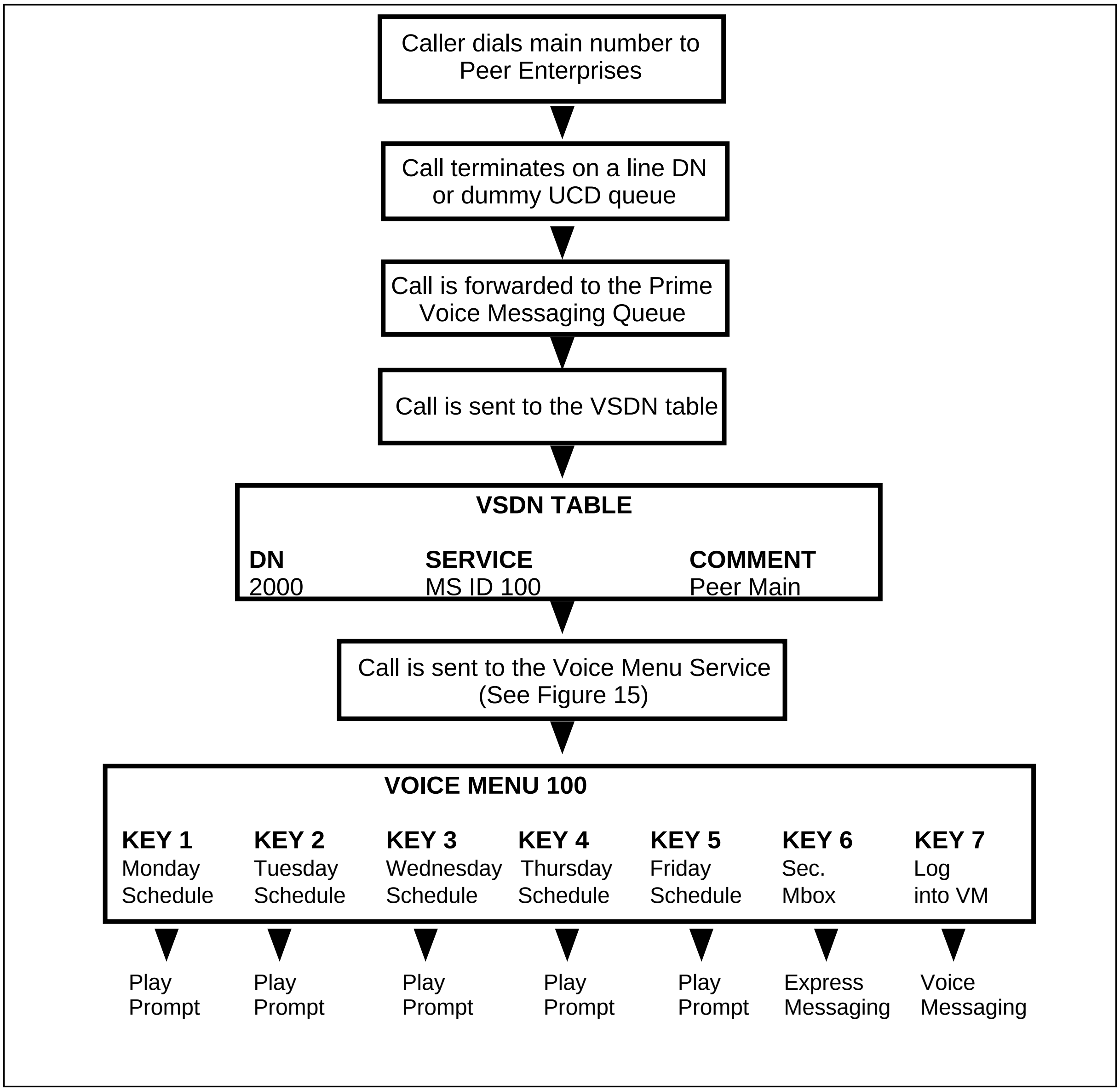


Figure B-15xxx
The Voice Menu Definition for multi-layer announcements

```
                    VS Config/Menu Applications Admin

  Add a Voice Menu Definition

* Choice of Menu Actions:
  AS Announcement Service    CL   Call                    RV   Call Revert DN
  DS Disconnect              EM   Express Messaging       GS   Greetings Service
  PP Play Prompt             RP   Repeat Menu Choices     MM   Return to Main Menu
  TS Thru-Dial Service       TD   Time-of-Day Control     TR   Transcription Service
  VF Voice Forms Service     MS   Voice Menu Service      VM   Voice Messaging

  Voice Menu ID:    100                    Title:   Peer Main

  Revert DN:        2398

  Access Password: 67902345          Update Password: 970098154

  Greeting Recorded (Voice):        Yes     Menu Choices Recorded (Voice): Yes

**Silent Disconnect:                [No]    Yes

  Language for Prompts: [AmericanEnglish]
                         CanadianFrench
                         AmericanSpanish
                         French

         Key             Action                         Comments
         1               PP                             Monday Schedule
         2               PP                             Tuesday Schedule
         3               PP                             Wednesday Schedule
         4               PP                             Thursday Schedule
         5               PP                             Friday Schedule
         6               EM   Mailbox ID: 2398
         7               VM
         8               PP
         9               PP
    Initial No Response  RP
    Delayed  Response    RV

  Select a softkey >

     Save          Cancel                         Voice
```

*   Some of these actions are feature-dependent and may not appear on your screen.

**This field is displayed if multiple languages are installed on your system.

Password protection

Recorded information that should be heard only by certain callers can be protected by having a password associated with the menu or announcement. Some examples would be salesmen receiving new product information, clients of a business receiving “special deals” information, and security personnel receiving confidential instructions.

Password protection can be put on a number of different items. A password can be defined for a particular menu, a specific announcement within a menu, or a thru-dial definition.

Example

A company wants a secure way to relay new product information to its sales people who are on the road. The announcement must be easily updated whenever necessary.

Setup

1 Announcement

“We have just added the new electronic model to our existing line two months before our competitor’s new product will be ready. The suggested retail price is being worked out. Call this announcement next weekend, or at the latest next Monday, for the list of suggested prices for the new model and replacement parts.”

2 Access Password

An Access Password is assigned and given to the sales people who will be calling this announcement.

3 Update Password

An Update Password is assigned so that the updating of this announcement can be delegated to the person available to record the announcement.

4 Revert DN

The revert DN is the number of someone able to give supplementary information to the sales people.

List of terms

68K card

68010 Processor card. Card with a 12Mhz 68010 processor, SCSI interface, serial port and the capability of addressing either 8 or 16 Mb and either 6 or 8 Mb of accessible RAM.

A

Analog

Pertains to representation by means of continuously variable physical quantities.

C

Card

A plug-in circuit pack containing components. In DMS, ”card” is the preferred term for a printed circuit pack or printed circuit board.

Central office (CO)

A switching office arranged for terminating subscriber lines and provided with switching equipment and trunks for establishing connections to and from other switching offices. Synonymous with class 5 office; end office; local office. *See* office classification.

Central processing unit (CPU)

A hardware entity, located in the central control complex frame, that contains the central data processor for the DMS-100 Family,

Centrex

Centralized PBX. A service that provides a Business telephone subscriber with direct inward dialing to extensions on the same system and direct outward dialing from all extensions. Centrex switching equipment is normally located at the central office, but may be located on the operating company client's premises.

Channel capacity

A measure of the maximum possible information rate through a channel, subject to specified constraints.

Circuit pack (CP)

In DMS-Supernode, consists of multi-layer PCB, through-hole electronic components, backpanel connector, faceplate, lock latches, and stiffeners.

CO

Central office

CPE

Customer Premises equipment.

Customer Premises Equipment (CPE)

Refers to equipment, such as ISDN terminals, that is located on the customer's premises.

D

Data

In translations, tables contain data. Each field or subfield has specific data values which are valid for that field. For example, a field called SECONDS may accept integer values from 0 through 60. A field called DAY may accept values of SUNDAY, MONDAY, TUESDAY. The set of all possible data values for a field is known as the *range* fo the field.

Datafill

In translations, datafill is the process of entering data into a table, for example, "I am going to datafill the table now". Datafill is also used as a synonym for data, for example, "The datafill in that table is incorrect".

DID

Direct inward dialing

Directory

In DMS, a software structure that may be used to look up, store, and delete symbols.

Directory number (DN)

The full complement of digits required to designate a subscriber's station within one NPA - usually a three-digit central office code followed by a four-digit station number.

Disk drive unit

Consists of a disk drive and a power-converter card installed in an input/output equipment frame.

DMS

Digital Multiplex System

DMS-Supernode

A central control complex for the DMS-100. the two major components of DMS-Supernode are the computing module and the message switch. Both are compatible with the current network module, the input/out controller, and the XMS-based peripheral modules.

DMS-100 family of switches

A family of digital multiplexed switch systems, which includes the following:

DMS*-100
Local switch

DMS*-200
Toll switch

DMS*-100/200
Switch of mixed function, in this case a combined local/toll switch. Other combinations are possible.

DMS*-250
Toll switch designed for private toll networks.

DMS*-300
Gateway switch

DMS-100* switching cluster
A DMS-100 host, up to eight large business remotes, and a centralized operation, administration, and maintenance application. Together these components operate and are maintained as a single switching center.

---

* Trademarks of Northern Telecom.

DMS-100* switching network
Multiple DMS-100 Family products that are maintained from a centralized operation, administration and maintenance application.

DN

Directory number

E

Error message
An indication that an error has been detected.

G-H

Ground start line
A line circuit arrangement in which dial-tone is sent in response to a ground signal on the ring conductor applied by the calling station or PBX. This differs from the more common loop start configuration, in which seizure is accomplished by bridging the tip and ring conductors.

Hundred call seconds (CCS)
Calculated by multiplying the average number of calls during busy hour by the average holding time in seconds, divided by 100. 36 CCS=1 Erlang.

I

IF

Interface (card)

Input/output (I/O)
Refers to a device or medium that is used to achieve a bi-directional exchange of data. Data exchange in the DMS-100 Family system is performed in accordance with the input/output message system.

Input/output device (IOD)
A hardware device that interprets input and formats output for human users or remote computes.

Integrated Business Network (IBN)

Now known as Meridian Digital Centrex. A special DMS business services package that utilizes the data-handling capabilities of a DMS-100 Family office to provide a centralized telephone exchange service. Many optional features are also available.

Integrated Services Digital Network (ISDN)

A set of standards proposed by the International Telegraph and Telephone Consultative Committee (CCITT) to establish compatibility between the telephone network and various data terminals and devices. ISDN provides a path for transmission of voice, data, and images.

I/O

Input/output

L

Line hunting

Procedure for searching a number of lines to find one that is idle. See Multi-line Hunt.

Link

- In DMS, a connection between any two nodes. *See* node.
- A four wire group of conductors providing transmit and receive paths for the serial speech or message data between components of DMS-100 Family systems. Speech links connect peripheral modules to the network modules. Message links connect network message controllers or input/output controllers to the central message controller.

Link protocol

A set of rules for data communication over a data link. Link protocols exist for transmission codes, transmission nodes, and for data control and recovery procedures.

M

Modem

Contraction of modulator/demodulator; a device that modulates and demodulates signals for transmission and reception, respectively, over communication facilities. A modem is used to permit digital signals to be sent out over analog lines. Synonymous with data set.

Module

- The basic building block of software structure. A module consists of interface and implementation sections.
- A discrete hardware package, designed for use in conjunction with other components.

MSP

Multi-server Processor

Multi-line Hunt

A service-related telephony feature that permits calls to a busy line be routed to other specified lines without assigning a directory number to each line. Refer to line hunting.

Multi-protocol controller (MPC)

A general-purpose data communications card that allows data communications between a DMS-100 Family switch and an external computer (between a central office billing computer and a DMS-100 Family switch, for example). The MPC card resides on the input/output controller shelf. The MPC card's protocol software is downloaded from the DMS-100 central processing unit and then supports software routines for data packet network communication.

Multi-server Processor

A node running multi-server programs in a multi-node environment, ie on the Service Peripheral Module.

N

Network

- An organization of stations capable of intercommunication but not necessarily on the same channel.
- Two or more interrelated circuits.
- A combination of terminals and circuits in which transmission facilities interconnect user stations directly.
- A combination of circuits and terminals serviced by a single switching or processing center.
- An interconnected group of computers or terminals.
- (NET) The network module frame of the DMS-100 Family system.

Node

The terminating point of a link. Node is a relative term; its meaning depends entirely on the context within which it is used. For example, a circuit may be a node in the context of another circuit within a module; the module itself may be a node in the context of another component of the network, and so forth.

Northern Telecom Practice

A document that contains descriptive information about the DMS-100 Family hardware and software modules, and performance oriented practices for testing and maintaining the system. NTPs are supplied as part of the standard documentation package provided to an operating company.

NTP

Northern Telecom Practice

O

Operating company

The owner/operator of a DMS switch.

P-Q

PBX

Private branch exchange

Peripheral Equipment (PE)

Equipment which works in conjunction with a communication system or a computer but is not part of it. In the DMS-100 Family of switches, it is a general term applied to peripheral modules.

Peripheral Module (PM)

A generic term referring to all hardware modules of the DMS-100 family systems that provide interfaces with external lines, trunk, or service facilities. PM contains peripheral processors which perform local routines, thus relieving the load on the central processor unit.

Plain ordinary telephone system (POTS)

POTS is an acronym used in the telephone industry to denote basic, conventional telephone services.

Port

In DMS, the point at which a speech or message link is connected to a peripheral module, network module, input/output controller, or central message controller.

Private branch exchange (PBX)

A private telephone exchange, either automatic or attendant-operated, serving extensions in an organization and providing access to the public network.

S

Service Order System (SERVORD)

A user interface used to change, add, or delete a subscriber line. Standard telephone industry command-format is used.

Service Peripheral Module (SPM)

A voice processing server used to provide voice messaging and related services for residential and business subscribers of DMS-100 or other central office switches.

SERVORD

Service Order System

Shelf

A container for drawers, cards, or both.

Signal Processing Node (SPN)

A node on the Service Peripheral Module that is used for signal processing.

Simplified message desk interface (SMDI)

An interface feature that enables a DMS-100 switch to communicate with a message desk. It provides the directory number of the called station, the calling station number (if available), and the reason for the call being forwarded to a message desk. In addition, it provides the message desk with the ability to activate or deactivate the message waiting indication for any station able to forward calls to the desk.

SMDI

Simplified message desk interface

SPM

Service Peripheral Module

SPN

Signal Processing Node

Subscriber

An individual user of a telephone station set that is connected to a DMS switch. Also known as end user.

T

Tape unit

See magnetic tape unit.

Telephony Interface Node (TIFN)

A node that is used to interface between incoming telephony lines and place the communications on the MM bus of the Service Peripheral Module.

Terminal

- The point of origination or termination in a communications network.
- Any device capable of sending and/or receiving information over a communication channel.
- Also, in DMS, the smallest unit of address space within the input/output system.

TIFN

Telephony Interface Node

Transition Module (TM)

A short circuit pack, based on the standard circuit pack. The TM carries the cable interfaces and/or local service functions such as local clock sources and bus terminations, located on the back of DMS-Supernode shelf.

Translations

Translations is the process the DMS-100 family of switches uses to determine the destination of a call based on the digits the caller dials and the capabilities available to the caller. It also allows the DMS software to recognize the hardware components fo the system.

T1

The standard 24-channel, 1.544 Mb/s pulse code modulation system as used in North America. This digital carrier carries a signal whose designation is DS1.

U-V

UCD

Uniform Call Distribution

Uniform Call Distribution (UCD)

A Meridian Digital Centrex feature which allows calls to be evenly distributed to a number of pre-designated stations known as UCD stations or UCD positions. This feature is used to queue incoming calls to the message desk.

Voice Processor-12 card

A twelve port card that is used in the Service Peripheral Module for voice processing.

VP12

Voice Processor-12 card

DMS-100 Family and SL-100

DMS VoiceMail

Administration Guide (for single-customer systems)

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