ReadiVoice®

Administration & Maintenance Guide

POLYCOM®

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About This Manual

This introduction provides a brief overview of the *ReadiVoice Administration & Maintenance Guide*, describes the conventions used in this manual, and explains how to get additional information or support.

Purpose

This manual is for administrators of a ReadiVoice system. It covers:

- Administering the ReadiVoice application. Describes using the Web-based System Administration interface to add, modify, and view the system and user data.
- Monitoring the ReadiVoice system. Describes using the monitoring tools in the System Administration interface to monitor the status of the system, usage levels, conference data, and operator data.
- Maintaining the ReadiVoice system. Includes routine maintenance tasks, backup and database maintenance procedures, and other maintenance tasks you may need to perform from time to time.
- Configuring the ReadiVoice system. Includes procedures for changing how the system works and what capabilities, features and options are enabled. Many of these tasks involve editing configuration files and working at the UNIX command prompt.
- Customizing and branding the ReadiVoice system. Describes customizing the system for your company and customizing how specific sets of users experience the system.
- Diagnosing and troubleshooting the ReadiVoice system. Describes tools and procedures for identifying problems with the system.

In addition, the appendices of this manual provide reference information about the system, including voice prompts and call flows, CDR (call detail record) data, and SNMP data.

Document Conventions

This document uses the following typographical conventions.

Typeface	Usage
bold	Names of fields, screens, windows, dialog boxes, and other user interface elements; for example: 1 Type the number into the **Phone Number** field and click **Dial**. 2 Click **Cancel** to close the dialog box.
italics	New terms, book titles, or emphasis; for example: According to the *VERITAS Cluster Server User Guide*, crash tolerant applications are sometimes referred to as *cluster friendly* applications.
`code`	Computer output, command references within text, and filenames; for example: Performs the initial configuration and reads the `.vcsrc` file
`code, bold`	Command line entries, for example: >> Type **`cp ../default_group.ini group.ini`**.
`code, bold & italics`	Command line variables, for example: >> Type **`cp ../default_group.ini group`*****`nn`*****`.ini`** replacing ***`nn`*** with the subscriber group number.
SMALL CAPS	Specific keys on the keyboard, for example: >> Move the cursor by pressing TAB or SHIFT+TAB.

Support

Recognizing that technology alone cannot solve today's complex challenges, Polycom Global Services provides the industry's best technical support staff and programs to let you concentrate on the task at hand. ReadiVoice users can select from a variety of support solutions to obtain the level of support that best meets their needs.

Before contacting your Polycom Global Services representative for technical assistance, gather as much information as possible about your situation. Any information you can provide helps us assess the problem and develop an appropriate solution.

Polycom Global Services Telephone and Email

If you have comments or questions about ReadiVoice or if you need technical assistance, contact:

- Polycom Global Services in U.S.A 800-827-7782
- Denver metro or outside U.S.A. 303-223-5223
- Email: techsupport@polycom.com

Polycom Global Services on the Web

For more information about Polycom Global Services, go to http://www.polycom.com and select the Support Solutions link under the Global Services category.

Polycom Technical Publications Department

If you have comments or questions about this or any other ReadiVoice documentation, contact:

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1

Introducing the ReadiVoice® System

This chapter offers a general overview of the ReadiVoice conferencing system and its features, functionality, and components.

What is the ReadiVoice System?

The ReadiVoice is the industry-leading, on-demand conferencing system. It provides an easy-to-use, reliable, scalable, and full-featured conferencing solution.

A ReadiVoice conference may contain up to 300 participants, including the conference leader(s).

How ReadiVoice Conferencing Works

The ReadiVoice system is *subscriber*-based. A subscriber is someone, such as a chairperson or conference leader, for whom you've set up conferencing access. Each subscriber has access to a specified number of conferencing ports and can use them at any time.

Through the *provisioning* process, you add subscribers to the ReadiVoice system's database. Each subscriber is given an *access phone number* for dialing into the system and a *subscriber password* that identifies the subscriber. An access number can be shared by many subscribers or private (assigned to only one subscriber). If the access number is shared, the subscriber is also given either:

- An *access code* that everyone uses to access the subscriber's conferences.
- A *participant password* that participants use to access the subscriber's conferences (the subscriber uses the subscriber password).

Each conferencing subscriber is a member of a *subscriber group*. Subscriber groups are logical groupings of related subscribers who share the same default settings and options (for instance, a specific prompt set). You can create groups for specific purposes or you can use one subscriber group for all subscribers.

Starting and Joining a Conference

Once provisioned, a subscriber can hold a conference at any time. No advance reservation is necessary or possible. Subscribers can start conferences from any touchtone telephone by dialing the access phone number and providing their access code (if any) and subscriber password when prompted.

Participants join a conference by dialing the access phone number and providing their access code or participant password. If the account permits, the subscriber can also specify a conference security code that participants must know to join the conference.

Controlling a Conference

A subscriber can have only one conference active at a time. During a conference, subscribers can control the conference and dial out to additional participants using:

Telephone touchtone (DTMF) commands — The ReadiVoice system supports a complete set of touchtone commands. Subscribers can use these to control the conference, enable or disable features, and dial out. Participants can use these to control their own lines and use features to which they've been given access.

The touchtone command set is completely configurable at the system level. In addition, you can enable or disable specific features at the subscriber group or individual subscriber level.

The Web-based Moderator interface — The Moderator lets your conferencing subscribers or their designated moderators set up, start, monitor, and control their conferences over the Internet. To log into the Moderator, users enter the subscriber's access phone number, access code (if any), and password.

The Moderator provides the conference control features that are enabled in the subscriber profile for the conference. Most commands available to the subscriber as touchtone commands are also available in the Moderator.

By default, conferences start when the subscriber arrives and end when the subscriber leaves. This can be modified as follows:

- If the Quick Start option is enabled, participants don't have to wait on hold for the subscriber. The conference starts when the first participant arrives.
- If Continuation is enabled, the conference doesn't end when the subscriber leaves. It continues until the last participant leaves.
- The system can be configured to end a conference (or call an operator) containing only one to three lines (configurable) after a specified delay. You can use this feature to handle problems such as someone forgetting to hang up a speaker phone in a conference room.

For more detailed information about how subscribers and participants interact with the system to set up, start, control, and join conferences, see the *ReadiVoice Subscriber Guide*.

Managing Your ReadiVoice System

Day-to-day management and operation of your ReadiVoice system is accomplished using *operator/maintenance stations,* which are typically standard PCs connected to your company's LAN (local area network) or WAN (wide area network). From these PCs, your company's staff can access the Web-based ReadiVoice applications for:

Operators — Respond to operator requests from subscribers (and, if authorized, participants) to resolve problems, answer questions, or provide other assistance.

Provisioners — Add, modify, and delete subscriber accounts.

System Administrators — Configure and maintain the ReadiVoice system, create new subscriber groups, and set up other internal users and their passwords. They can also perform the provisioning functions.

To use a Web-based ReadiVoice application, users launch a supported browser (Microsoft Internet Explorer version 5.5 or later), point it to the correct Internet address (URL) for the application, and log in when prompted.

The Operator application and some parts of the Administration interface also require a Java virtual machine (plug-in), which the user is prompted to download if it isn't already installed. Voyant has certified these Java applets with the Sun Microsystems Java Runtime Engine (JRE) 1.3.1 and 1.4.2, on Microsoft Windows 2000 and Microsoft Windows XP.

Note: These applets should work on other platforms and operating systems and with newer versions of the JRE, but that depends on proper implementation by the operating system and Java virtual machine. Polycom hasn't tested other possible combinations and can't be responsible for implementation or compatibility issues beyond our control.

Operator Interface

The ReadiVoice Operator interface lets operators monitor the conferences on the system, answer operator requests from their subscribers and participants, and perform various tasks to assist users.

Note: The ReadiVoice system doesn't *require* operators. Providing operator services is entirely optional.

Operators may have access to all conferences or they may be limited to the conferences of a specific subscriber group or groups, depending on their login name and the system configuration.

When someone requests an operator, the system sends an audible and visual alert to the available operators. An operator can answer the oldest request in the queue or select a specific request from a list.

When operators answer a request or select a conference to monitor, the system registers them for the conference and displays all the available information about the conference and its participants.

If an operator's duties include provisioning as well as answering operator requests, the operator can switch to the Provisioning interface by clicking the **Provisioning** button on the **Operator** page.

For more information about the Operator interface and operator tasks, see the *ReadiVoice Operator Guide*.

Provisioning Interface

The ReadiVoice Provisioning interface lets provisioners create and modify ReadiVoice subscriber accounts. A subscriber account authorizes someone to use your ReadiVoice system.

Note: We recommend automating your initial subscriber provisioning. Contact Polycom for help with this. The ReadiVoice system's Provisioning interface is available for manual provisioning and for viewing or modifying subscriber account information.

To set up an account for a customer, a provisioner enters subscriber and billing information and defines the subscriber's conference access and features.

All subscribers belong to a *subscriber group*. You can use one subscriber group for everyone, or you can have multiple subscriber groups, each used for a specific company, access method, or other affinity group. A subscriber group can have its own default voice prompt set, account settings, and number group (controlling access numbers and routing).

Like operators, provisioners may have access to all subscriber groups or they may be limited to a specific subscriber group, depending on their login name and the system configuration.

The Provisioning interface lets provisioners create new subscriber accounts or search for an existing account and then edit or disable it.

For more information about the Provisioning interface and provisioning tasks, see the *ReadiVoice Provisioning Guide*.

Administration Interface

The ReadiVoice Administration interface lets administrators monitor the system, create new subscriber groups, customize the interfaces, create and maintain user logins, set default parameters, and configure bridges.

This manual describes the Administration interface and administrative tasks.

Components of the ReadiVoice System

A ReadiVoice system includes:

- A Conference Allocation and Control System (CACS) server.
- Up to twelve InnoVox media servers (conferencing bridges), providing up to 5,760 ports in a single system, or one InnoVox media server, providing 4032 ports in a single 13U bridge cabinet.
- An optional boot server from which the bridges boot.

The specific configuration of a ReadiVoice system depends on its capacity (number of ports), type of media server(s), and options.

The CACS and boot server may be the same server. All system components mount into a standard 19-inch rack.

Two types of ReadiVoice systems are available. There are both hardware and software differences between the two types, although the core ReadiVoice software is the same (merely configured differently). The two types are for different network environments:

ReadiVoice-PSTN uses the Public Switched Telephone Network (PSTN). The bridge or bridges make circuit-switched connections over standard digital telephony spans.

ReadiVoice-IP uses Voice Over IP (VOIP). The bridge makes packet-switched Internet Protocol (IP) connections over Ethernet.

ReadiVoice-PSTN Systems

The PSTN version of ReadiVoice has two call routing configurations:

Fixed Access — Each access telephone number terminates on a specific bridge in the system. This configuration is called a *non-routed* system.

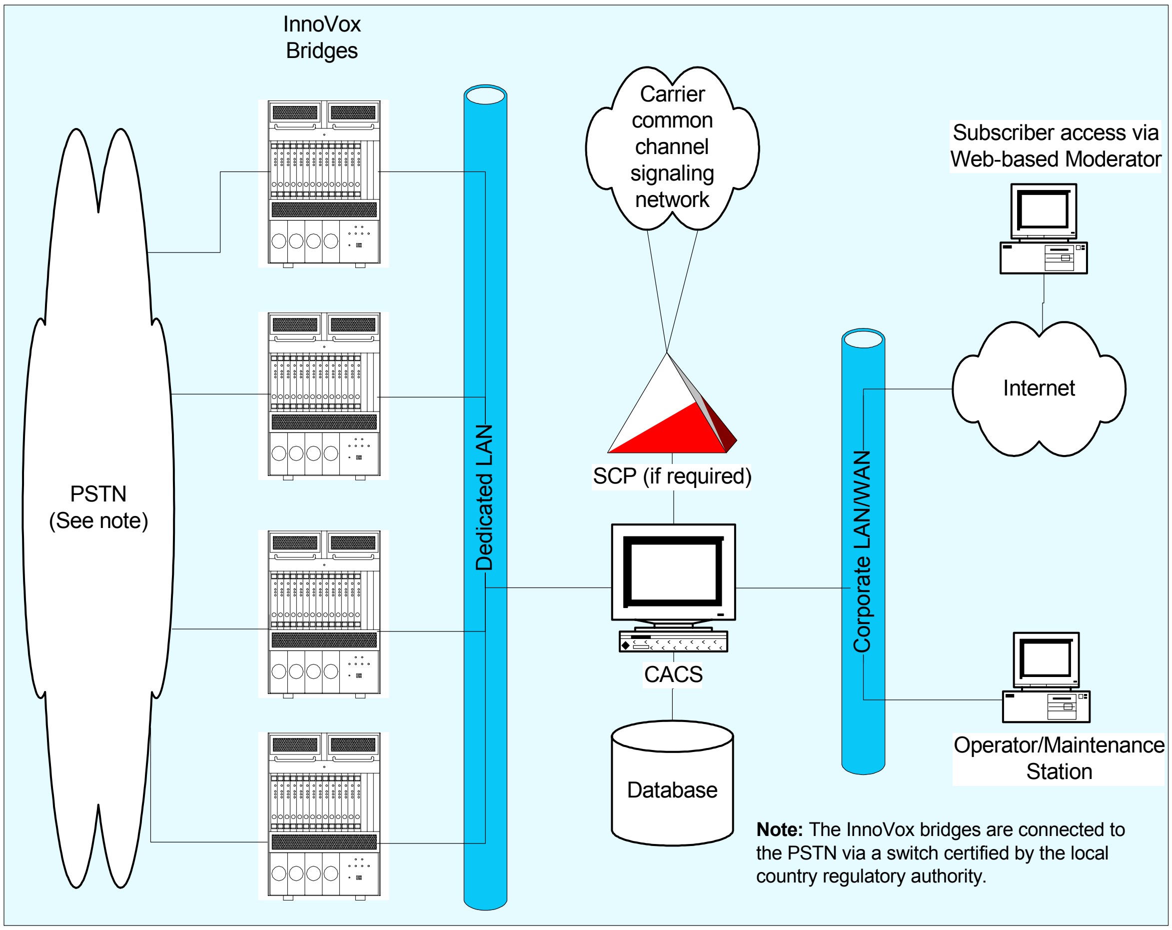
Intelligent Network Call Routing (INCR) — Access telephone numbers aren't assigned to specific bridges. Instead, the ReadiVoice system uses a carrier common channel signaling network (such as SS7 or Tollfree Gateway) to dynamically route calls among available bridges. This configuration is called a *routed system*. A routed system makes more efficient use of the bridges because you can distribute the conferencing load and specify bridge availability and priority.

Figure 1-1 provides a high-level view of the major components in a routed ReadiVoice-PSTN system attached to a carrier common channel signaling network. A fixed access system is identical, but without the Service Control Point (SCP) connecting it to the carrier common channel signaling network.

The SCP may or may not be required for a routed system, depending on your carrier.

The SCP (if required), subscriber database, and Web server can physically reside on the CACS server. In large systems, however, these components may be distributed among multiple servers in order to distribute the processing load.

Figure 1-1 *ReadiVoice-PSTN system diagram*



When a caller dials a ReadiVoice-PSTN subscriber's access phone number on a typical routed ReadiVoice system:

1 The originating central office switch recognizes the number as requiring carrier common channel signaling network handling.

2 The carrier switch sends a query through the carrier common channel signaling network to the ReadiVoice CACS, which allocates bridge resources to that subscriber according to his or her subscriber account settings.

3 The CACS returns a routing solution through the carrier common channel signaling network to the originating carrier switch. It also notifies the bridge of the upcoming call and the associated account profile and features.

4 The carrier switch uses the routing solution to route the call to the InnoVox bridge selected by the CACS call router.

   Figure 1-2 summarizes this INCR signaling flow.

5 When the call reaches the bridge, the ReadiVoice system places it into an interactive call flow script to guide the caller into the conference. In a fixed access system, the script collects the access code (if needed). In all systems, the script handles subscriber identification and collection of the optional conference security code. Non-subscribers are put on hold until the subscriber arrives, enters the subscriber password, and starts the conference.

6 If the account permits, the call flow script gives the subscriber an opportunity to review and change account options prior to starting the conference.

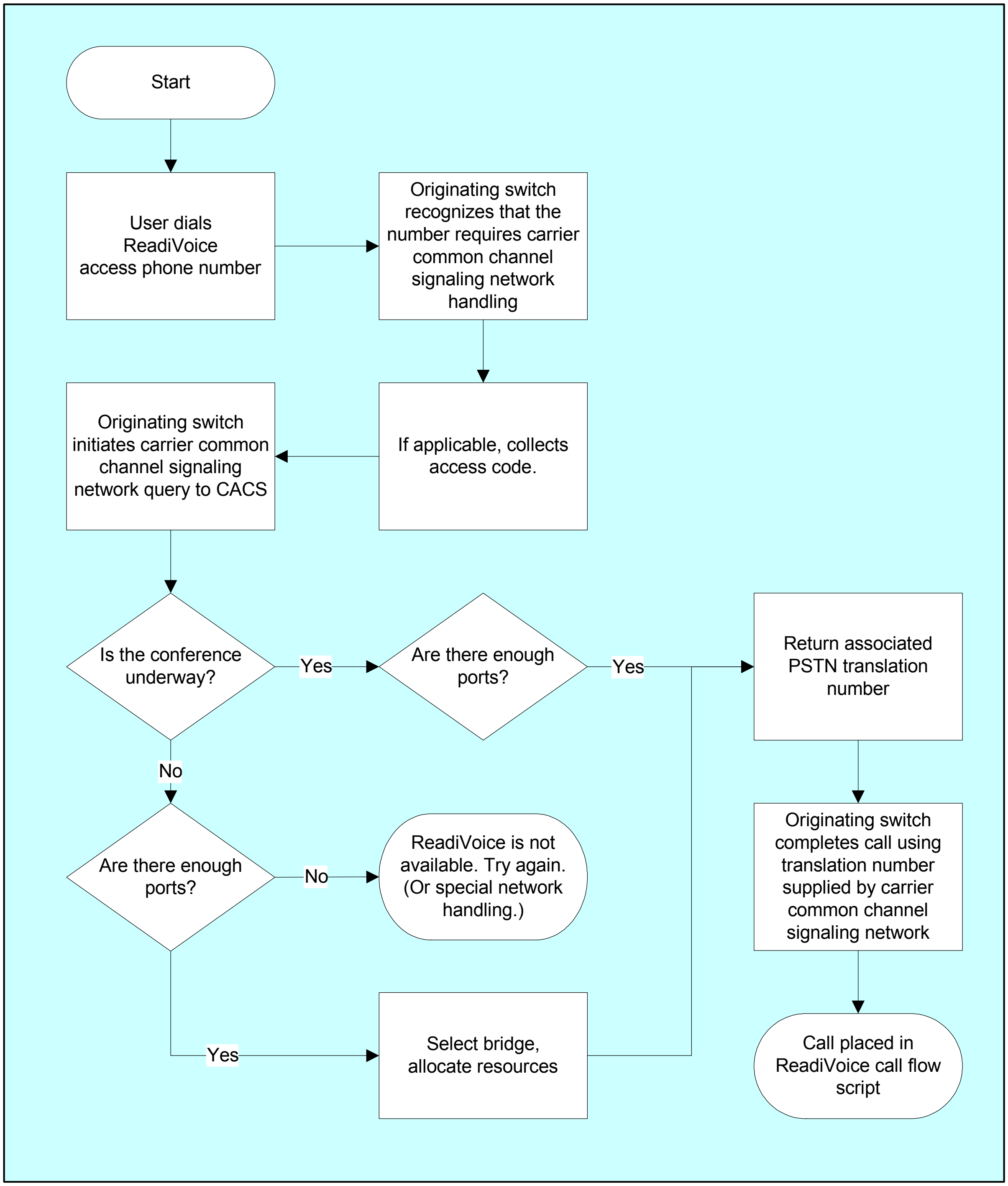
7 Once the subscriber starts the conference, participants are prompted to enter the conference security code (if applicable) and then placed into the conference.

Figure 1-4 on page 13 summarizes the ReadiVoice basic call flow after the call reaches the bridge or on a non-routed system.

If the Quick Start feature is enabled, participants don't have to wait for the subscriber. They're placed directly into conference instead of on hold. If the Roll Call feature is enabled, callers are prompted to record their name before entering the conference.

Callers who enter the subscriber or participant password or the conference security code incorrectly three times are played the appropriate message and then either sent to an operator (if available) or disconnected, depending on how your system is configured.

Figure 1-2 *ReadiVoice-PSTN INCR signaling flow*

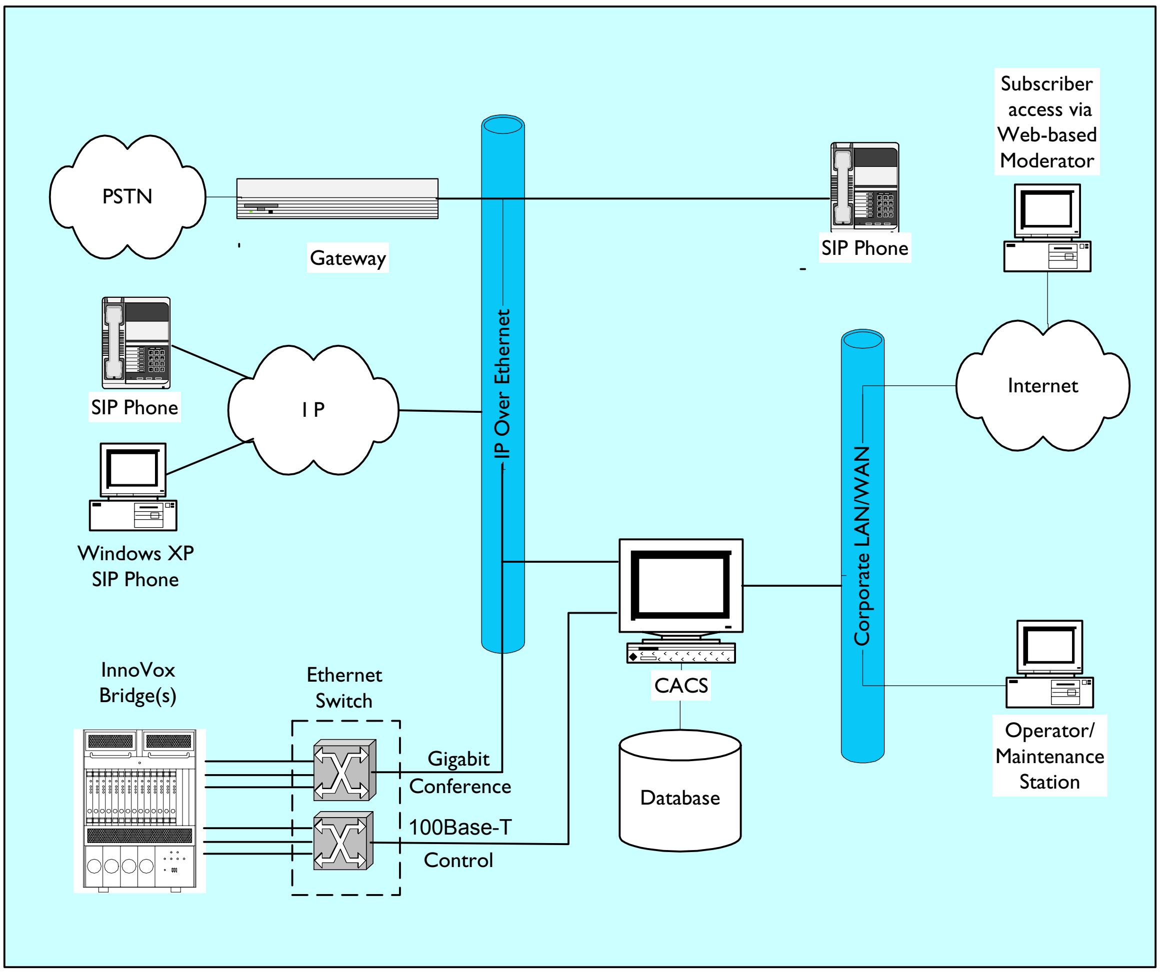


ReadiVoice-IP Systems

All ReadiVoice-IP systems are routed. That is, the CACS uses SIP (Session Initiation Protocol) signaling over IP to route calls dynamically among available bridges.

Figure 1-3 provides a high level view of the components in a ReadiVoice-IP system.

Figure 1-3 *ReadiVoice-IP system diagram*



ReadiVoice-IP Call Processing

When a caller dials a ReadiVoice-IP subscriber's access phone number on a typical ReadiVoice system:

1 The SIP Invite message from the caller's SIP user agent (perhaps a SIP telephone) reaches the call-control portion of the CACS. If the SIP invite contains no access code:

   a The CACS directs the call to a bridge with a free port and runs a bridge script that prompts the caller for the access code.

   b The bridge returns that access code to the CACS.

2 The CACS uses the access code to identify the subscriber account. Its call router allocates bridge resources according to the service description of the subscriber just identified.

3 The CACS connects the call to the bridge just chosen.

4 When the call reaches the bridge, an interactive call flow script guides the caller into the conference. The script handles subscriber identification and collection of the optional conference security code. Non-subscribers are put on hold until the subscriber arrives, enters the subscriber password, and starts the conference.

5 If the account permits, the call flow script gives the subscriber an opportunity to review and change account options prior to starting the conference.

6 Once the subscriber starts the conference, participants are prompted to enter the conference security code (if applicable) and then placed into the conference.

Figure 1-4 on page 13 summarizes the ReadiVoice basic call flow after the call reaches the bridge.

If the Quick Start feature is enabled, participants don't have to wait for the subscriber. They're placed directly into conference instead of on hold. If the Roll Call feature is enabled, callers are prompted to record their name before entering the conference.

"Dialing" in ReadiVoice-IP Systems

For the sake of simplicity, we refer to "dialed phone numbers" throughout this document. Actually, in IP telephony, end points are identified by a Universal Resource Identifier (URI).

This URI resembles an email address: phonecontact@domain. Domain is pretty much what one would expect: something.com or something.org, for example. Phonecontact can be anything that identifies a device known to the domain, such as a name, a 10-digit phone number, or perhaps an extension number.

The ReadiVoice-IP system uses the Session Initiation Protocol (SIP) to communicate with other IP telephony equipment. It sends out formal SIP URIs, but users can "dial" any of the following:

Formal SIP URI

The URI described above preceded by the name of the protocol and a colon. For instance:

SIP:3032235000@voyanttech.com
SIP:brent@voyanttech.com
SIP:5000@voyanttech.com

Simplified URI

The URI with the protocol name omitted (similar to omitting the "http:" in a Web address). For instance:

3032235000@voyanttech.com
brent@voyanttech.com
5000@voyanttech.com

URI Without Domain

If your system is configured with a default domain and port, it routes addresses that don't specify a domain to the default domain and port. For instance:

3032235000
brent
5000

Since most of the world is still using the PSTN and ordinary phone numbers, your ReadiVoice-IP system undoubtedly sits behind one or more *gateways* to the circuit-switched network. When your system is given an ordinary phone number with no domain specified (such as the example 3032235000 above), it routes the call to the gateway (or to a router that sends it to the appropriate gateway).

ReadiVoice Call Flow

The basic call flow is the same for ReadiVoice-PSTN and ReadiVoice-IP (see Figure 1-4). Various system-level and subscriber-level configurations and settings affect the basic call flow, however. The sections that follow describe some of these call flow differences.

How Quick-Start and Conference Continuation Affect Call Flow

By default, conferences start when the subscriber arrives and end when the subscriber leaves. Two options can change this:

Quick Start

The conference starts as soon as the first participant arrives. Participants can speak to one another prior to the arrival of the subscriber or hold a conference without the subscriber. Quick Start is useful for certain types of users, such as disaster management groups and others who need to meet without relying on any one individual to start the meeting.

Subscribers should be aware that Quick Start conferences are less secure than conferences requiring a subscriber password.

Conference Continuation

The conference can continue after the subscriber disconnects. In this case, the conference ends when the last participant disconnects.

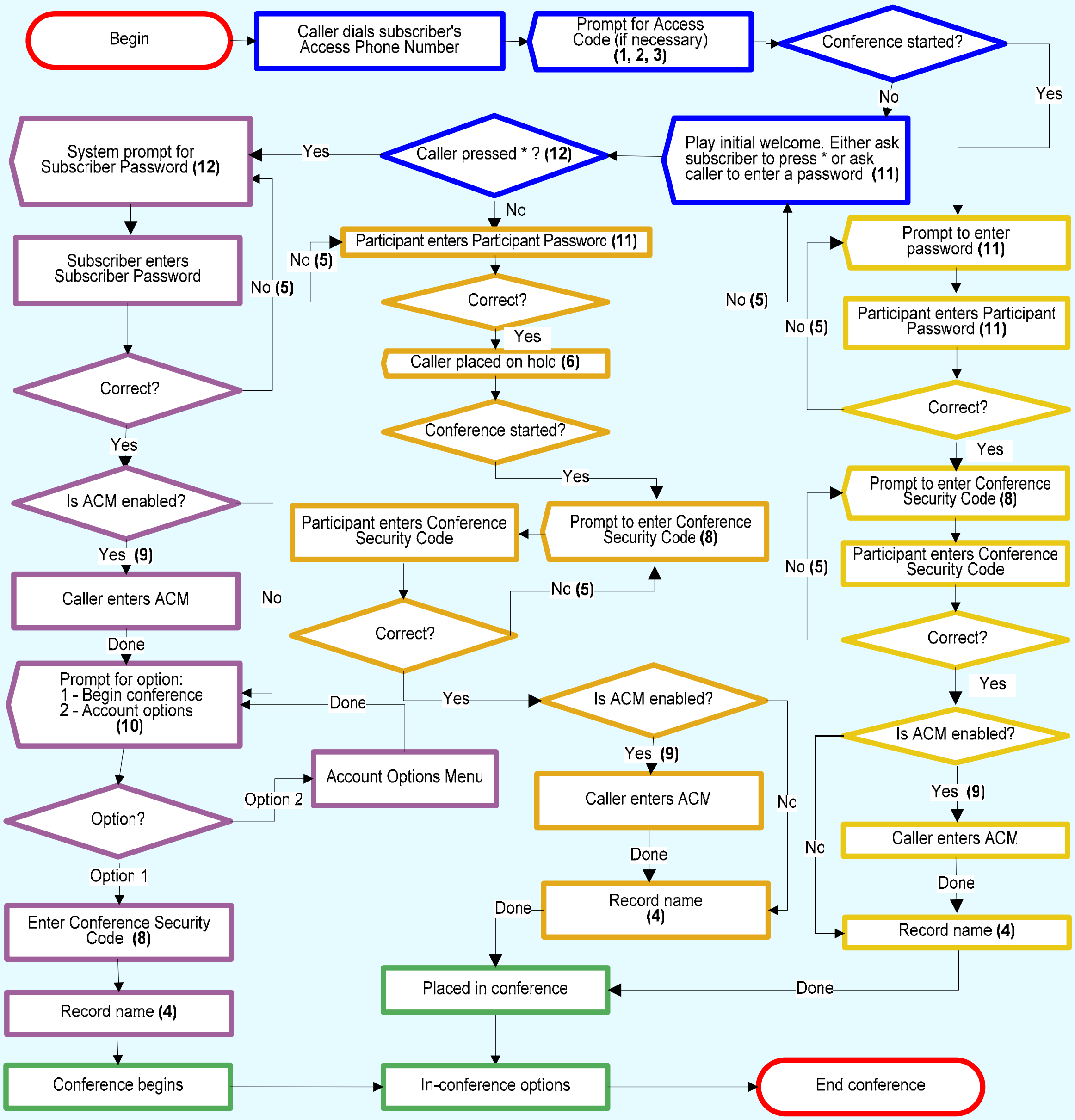
If the subscriber's account permits, the subscriber can turn Conference Continuation on or off for each conference. If the subscriber's account has Auto Continuation enabled, all the subscriber's conferences start with Conference Continuation turned on.

How One-Click Conference Affects Call Flow

In a One-Click Conference, participants click on a link or icon to enter the conference. The ReadiVoice system identifies the conference from the link's URL and returns a Web page asking for the participant's phone number. When the system gets this number, it calls and puts the connection in conference (or on music hold, if the subscriber isn't present and Quick Start is off). The subscriber can use the same link, since the Web page provides a checkbox to indicate subscriber and a field for entering the subscriber password.

The Provisioning application generates the One-Click Conference link for a new subscriber account. If the provisioners have email capability, they can paste the link into an email message and send it to the subscriber for distribution to participants.

Figure 1-4 *ReadiVoice call flow*



(1) In an Intelligent Network Call Routing (INCR) system, the carrier's network takes care of this.
(2) If the Access Phone Number is shared, an Access Code is required to identify the specific subscriber.
(3) Depending on your system configuration, additional Access Code prompts may be possible.
(4) If account has Roll Call feature enabled.
(5) After three incorrect attempts, caller is routed to an operator (if available) or played a message and disconnected.
(6) For Quick Start conferences, participants bypass hold and are placed directly into conference.
(7) The caller remains on hold until the subscriber arrives and starts the conference.
(8) Subscription setting determines whether Conference Security Passcode is mandatory, optional, or unavailable. If optional, the subscriber chooses whether to use a Conference Security Code for each conference.
(9) ACM (Application Control Mode), if enabled, removes callers from the call flow for an interactive session with an external application.
(10) Subscription setting determines whether subscriber has access to Account Options Menu.
(11) There is no request to press * if a Participant Password is enabled and specified.
(12) Skip if Prompt For Subscriber is set to Off; subscriber is not asked to press *, and, instead, a generic password request is played.

How A Two-Password Configuration Affects Call Flow

In the traditional ReadiVoice call flows, the system:

- Prompts all callers for an access code (unless the subscriber has a private access number) to identify the conference.
- Prompts callers to identify themselves as subscribers by pressing *.
- Prompts the subscribers for their subscriber password.

In two-password call flows, no separate access code is used; both subscribers and participants are given passwords. The system:

- Prompts all callers for their password.
- Uses it both to identify the conference and to distinguish between subscribers and ordinary participants.

With two passwords, subscriber identification occurs without any additional prompting by the system. This provides a shorter call flow.

How IP Tributaries Affect ReadiVoice-IP Call Flow

ReadiVoice-IP systems can be configured to enable the IP Tributaries feature. To use this feature, callers must be authenticated by some means outside of the ReadiVoice system, and the SIP `INVITE` message that they send to access the system must include the identifying information.

Each piece of information contained in the SIP `INVITE` message takes the place of information that the caller would otherwise be prompted to enter. For instance:

- If the SIP `INVITE` message includes an access code (traditional call flow), then the system can identify the conference from the invite and only needs to prompt for subscriber identification.
- If the `INVITE` message includes both the access code and subscriber password (traditional call flow) or either a participant password or a subscriber password (two-password call flow), then the system can identify both the conference and the caller's role and doesn't need to prompt the caller for any information.

  For security, the authentication mechanism can encrypt the passwords.

For more information on this feature, see Appendix D.

2

Administering the ReadiVoice System

This chapter describes how to use the ReadiVoice system's Web-based System Administration interface to add, modify, and view the system and user data. It covers:

Some of the functions available in the System Administration interface are described elsewhere:

- The changes you can make on the **System Configuration** page are described in Chapter 5, "Configuring the ReadiVoice System."
- The various monitoring functions available in the System Administration interface are described in Chapter 3, "Monitoring the ReadiVoice System."
- Installing multiple prompt sets and setting them up in the System Administration interface are described in Chapter 7, "Customizing & Branding Your ReadiVoice System."
- Using the **Critical Logs** page to help diagnose problems is described in Chapter 6, "Diagnostics and Troubleshooting."

Opening ReadiVoice System Administration

You can access the ReadiVoice System Administration interface from any computer that can connect to the ReadiVoice system's Web server and has a compatible Web browser (Microsoft Internet Explorer, version 5.5 or later, is compatible with all ReadiVoice interfaces).

Access is restricted to authorized users. You must know the correct user name and password.

To access the System Administration pages:

1 Do one of the following:

- Point your browser to the internal user home page (for example, http://www.rvoice.com/index2.html). Then click the **System Administration** link in the navigation bar on the right.
- Point your browser directly to the **System Administration** page (for example, https://www.rvoice.com/provisioning/Adminframe.html).

The **System Administration** home page appears (Figure 2-1). A navigation bar on the right provides links to pages for performing various administrative functions.

2 In the navigation bar, under the **System Administration** heading, click the link for the function you want.

All the **System Administration** links are restricted to authorized users. If you aren't already logged in as an authorized user, a login dialog box appears (Figure 2-2).

***Figure 2-1 System Administration** home page*

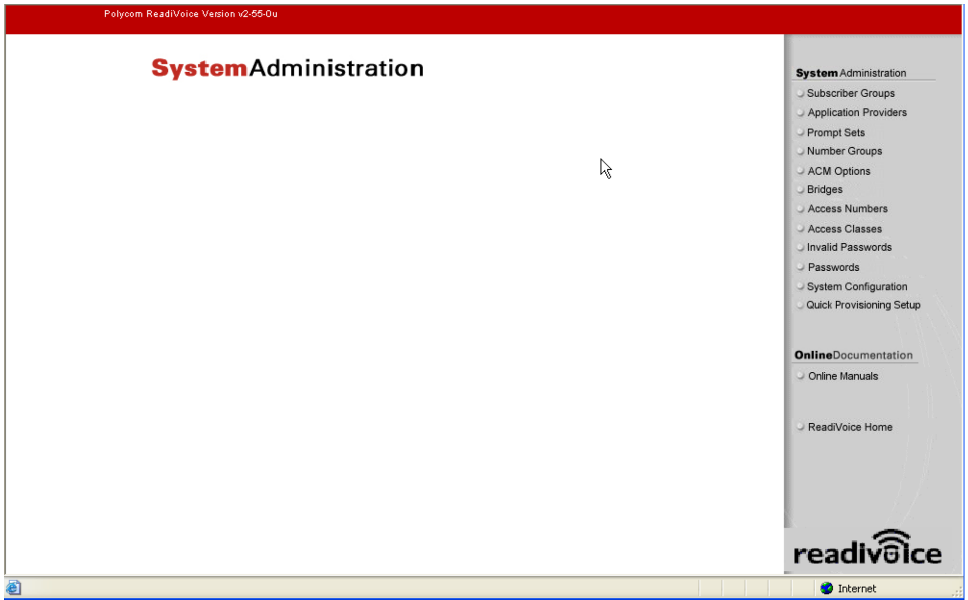
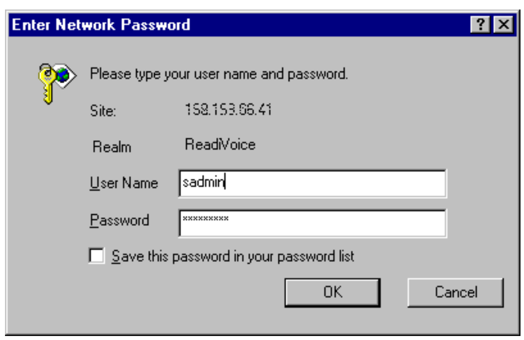


***Figure 2-2 Login** dialog box*



3 Enter a user name and password authorized for Administration access. Then click **OK**.

Maintaining Bridge Information

When Polycom installs your ReadiVoice system, we configure the application with the correct bridge information. If any of this information changes, you must update the system's bridge table.

Caution! Contact your Polycom Global Services representative before making any changes to your system's bridge configuration.

Adding a New Bridge

Installing a bridge (media server) involves:

- Physically installing the hardware and making all cable connections.
- Running the creator scripts to create the configuration files for the bridge.
- Using the `vbootptool` shell script to configure VBootP to bootstrap the bridge.
- Adding information about the bridge to the ReadiVoice database so that the ReadiVoice application is aware of its presence and able to use it.

This section describes only the last of these tasks. The prerequisite installation and configuration tasks are covered in the appropriate installation manual for the system.

This manual includes information about adding bridges for reference purposes; you shouldn't need it. When you buy a new bridge from Polycom, a field engineer installs the bridge and configures your system.

To add a new bridge to the ReadiVoice system:

1. In the **System Administration** navigation bar, click **Bridges**.

   The **Bridges** page appears (Figure 2-3 on page 20).

2. In the **Add Bridge** section at the bottom, if the new bridge's ID number isn't in the **Bridges** list, stop here.

   The installation of the bridge and VBootP configuration with the `vbootptool` shell script are incomplete.

   – Make sure that the `vbootptool` shell script was run.

   – Verify the VBootP configuration.

   – Make sure the bridge is powered on and properly bootstrapped fromVBootP.

3. In the **Add Bridge** section at the bottom, if the ID number for the new bridge is in the **Bridges** list, select it.

4 In the **Add Bridge** section at the bottom, complete the following fields:

Reserve Ports — Enter the number of ports you want to reserve for operator voice paths. These ports are not available for conferencing.

Trans DNIS Length — For IP systems, this is the number of digits in each translation number. For PSTN systems, enter the number of *DNIS* (dialed number identification service) digits the network delivers to the bridge. If the network delivers non-delimited or single-delimited ANI/DNIS, the `ive.ini` file also specifies a DNIS length, and the setting here must match that one.

Depending on the system's translation number type (see "Changing System Configuration Settings" on page 118), one of the following may also apply:

- If **Translation Number Type** is set to **Random Translation Number**, the number of digits must exactly equal the length of the translation numbers (not including the prefix, if any) assigned to the bridge (see "Working with Translation Numbers" on page 22). To change **Trans DNIS Length** to a different length, remove the translation numbers first.
- If **Translation Number Type** is set to **Fixed Translation Number**, the number of digits must exactly equal the number of digits in each subscriber's `ExternalId` field. If you change **Trans DNIS Length** to a different length, a warning message tells you that you must update your subscribers' external IDs.
- If **Translation Number Type** is set to **Fixed Translation Number with 3-digit code**, the number of digits must be exactly three more than the number of digits in each subscriber's `ExternalId` field. If you change **Trans DNIS Length** to a different length, a warning message tells you that you must update your subscribers' external IDs.

Hrt Beat Interval — Enter the interval (in seconds) between *heartbeat* messages from the bridge to the Conference Allocation and Control System (CACS) server. The heartbeat message confirms that the bridge is available and can communicate with the server. One second is generally considered a reasonable interval.

Max Missed Hrt Beats — Enter the number of heartbeats that the CACS can fail to receive before triggering an alarm. Eight is generally considered a reasonable number.

Bridge Status — Select **Busyout** to hold the bridge back from use until you've completed the bridge configuration.

5 Click **Add Bridge**.

The system provides a link to the **Translation Numbers** page.

6 Click the link, and then add translation numbers to the bridge as described on page 23.

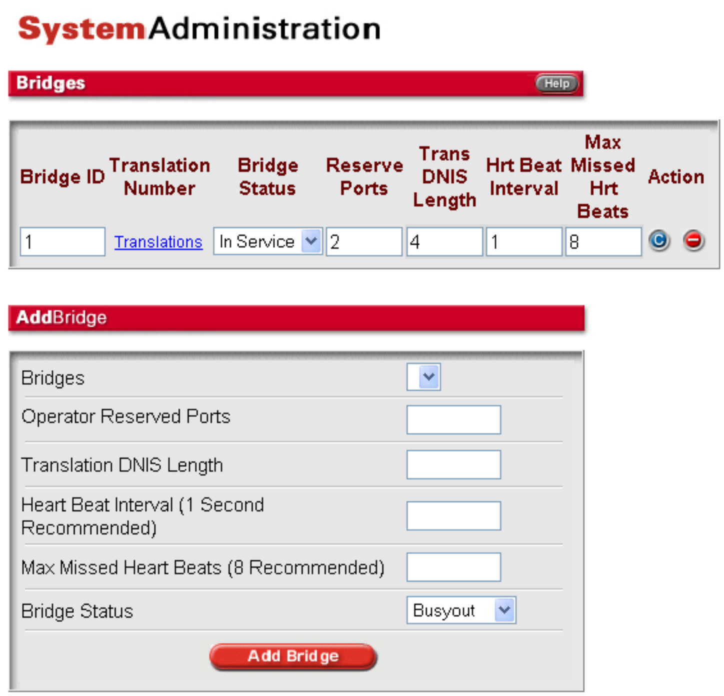
The system confirms that the translation numbers have been added.

7 In the navigation bar, click **Bridges**.

8 To make the bridge available for service, select **In Service** in its **Bridge Status** field and click the **Commit** button (blue "c").

The system confirms that the bridge's status has been changed and provides a link back to the **Bridges** page.

Figure 2-3 ***Bridges** page*



Modifying an Existing Bridge

1 In the **System Administration** navigation bar, click **Bridges**.

The **Bridges** page appears (Figure 2-3).

2 To change a bridge's status from **In Service** to **Busyout** or vice versa, select the status you want in the **Bridge Status** field.

3 To change a bridge's number of reserved ports, DNIS length, or heartbeat settings, edit the appropriate field.

If this is a routed system, you can't change the DNIS length to something different than the length of the bridge's translation numbers. If you must make this change, remove the translation numbers first (see "Changing the Translation Numbers Assigned to a Bridge" on page 24).

Caution!

> The value in the Bridge ID field must be unique and must correspond to the bridge's appBridgeID value in the vbootp.db file. If the bridge was properly configured in VBootP, its Bridge ID was set correctly when it was added to the ReadiVoice database, and it shouldn't be changed.

4. To save your changes, click the **Commit** button (blue "c").

   The system confirms that the bridge has been modified and provides a link back to the **Bridges** page.

To change a bridge's translation numbers, follow the procedure in "Changing the Translation Numbers Assigned to a Bridge" on page 24.

Deleting a Bridge

Before deleting a bridge, remove its translation number assignments. Then remove the bridge from any bridge groups to which it belongs.

To remove a bridge from the ReadiVoice application:

1. In the navigation bar, click **Bridges**.

   The **Bridges** page appears (Figure 2-3 on page 20).

2. Find the bridge you want to delete and click its Delete button (red "–").

   If the bridge has IP addresses or translation numbers associated with it or belongs to a bridge group, the system warns you of this and doesn't delete the bridge. Otherwise, it confirms deleting the bridge.

Working with Translation Numbers

A *translation number* is the telephone network's translation of the dialed telephone number into a number used to route the call:

- For a routed PSTN system, your carrier provides a block or blocks of translation numbers. Depending on the translation number type (see "Changing System Configuration Settings" on page 118), your system may randomly choose a number from this block, or it may use a subscriber-specific fixed translation number. In either case:
  - **a** When someone dials a ReadiVoice access number, the originating switch queries the CACS, which determines who the subscriber is and returns a translation number (including prefix) to tell the telephone network how to route the call.
  - **b** When the switch routes the call to the bridge, it sends the translation number, with or without prefix, as DNIS (dialed number identification system) digits.
  - **c** The bridge sends the DNIS to the CACS, which uses the last *n* digits (where *n* is the **Trans DNIS Length**) to identify the conference.
- For a non-routed system or an IP system, you must enter a block of dummy translation numbers. The number of digits must be the number specified on the Bridges page as **Trans DNIS Length** (see "Maintaining Bridge Information" on page 18).

In either type of system, each conference requires one translation number for each different access number used to reach the conference (see "Working with Access Classes" on page 33).

The maximum number of conferences is always the number of ports divided by three. Therefore, the maximum number of translation numbers a bridge might need is the smaller of:

- The number of access classes times one-third the number of ports.
- The number of ports.

Table 2-1 shows the results of this calculation for 480-port and 4032-port bridges.

Table 2-1 *Maximum number of translation numbers needed per bridge*

InnoVox 480 with 480 ports (max. 160 conferences)		InnoVox 4000 with 4032 ports (max. 1344 conferences)	
Access classes	**Translation numbers**	**Access classes**	**Translation numbers**
1	160	1	1344
2	320	2	2688
3+	480	3+	4032

These numbers are the worst-case scenario: each conference contains three lines, and each line used a different access nmber.

You may be able to reduce these numbers further, especially if most of your subscribers use a single access number from your primary access class, and the other access classes are used rarely. But, do so with caution. If a bridge runs out of translation numbers, no more calls can connect to it, even though ports are available.

Caution! Don't add or change translation number assignments while conferences are under way! Perform the following procedures after operational hours.

Adding Translation Numbers to a Bridge

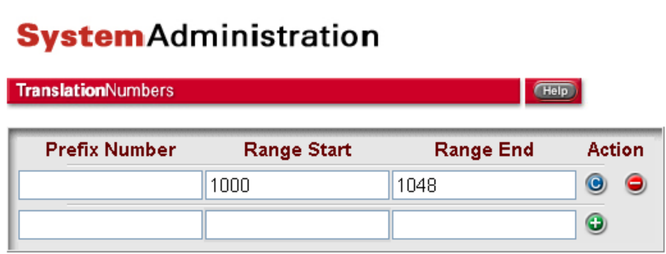
1 In the navigation bar, click **Bridges**.

The **Bridges** page appears (Figure 2-3 on page 20).

2 Find the bridge to which you want to assign translation numbers and click its **Translations** link.

The **Translation Numbers** page appears (Figure 2-4). If there are no translation numbers assigned to this bridge, the page contains only a blank row with an **Add** button (green "+") at the right.

Figure 2-4 ***Translation Numbers*** *page*



3 In the **Prefix Number** field, enter the prefix digits (the portion that the switch strips off before sending DNIS digits) of the number block you're assigning. Enter only the digits (no punctuation).

In a non-routed or IP system, leave the prefix field blank.

4 In the **Range Start** field, enter the number that begins the block of translation numbers. For instance, if your system receives only four DNIS digits from the switch, this might be **1000**.

5 In the **Range End** field, enter the number that ends the block of translation numbers. For instance, if your system receives only four DNIS digits from the switch, this might be **1479**.

6 Click the **Add** button.

The system assigns the translation numbers to the bridge. For a large block, this may take several minutes. When finished, it confirms that the translation numbers have been added and provides a link back to the **Translation Numbers** page.

7 Click the link to return to the **Translation Numbers** page.

The new row appears in the list. A blank row below lets you add additional blocks of numbers.

Repeat the above procedure to add another block of numbers.

Changing the Translation Numbers Assigned to a Bridge

1 In the navigation bar, click **Bridges**.

The **Bridges** page appears (Figure 2-3 on page 20).

2 Find the bridge whose translation numbers you want to change and click its **Translations** link.

The **Translation Numbers** page appears (Figure 2-4 on page 23). It contains a row for each block of contiguous translation numbers assigned to the bridge.

3 To change a block of translation numbers, edit the fields that need to be changed and click the row's **Commit** button (blue "c").

The system implements the changes you made. Depending on the extent of the change, this may take several minutes. When finished, it confirms the change.

4 To remove a block of translation numbers, click the row's **Delete** button (red "–").

The system confirms the change.

Working with Bridge Groups and Routing Lists

Bridge groups and routing lists apply only to routed systems (PSTN or IP). If you have a non-routed system, the access phone number determines which bridge receives a call, and this section doesn't apply to you.

In a routed system, bridge groups and routing lists control the routing of calls among bridges. Each subscriber is assigned to a routing list (by default, the one assigned to the number group, but provisioners can change this). The routing list specifies which bridge or bridges the subscriber's conferences can use and the priority order of those bridges.

Setting up bridge routing involves:

- Setting up *bridge groups,* each containing one or more bridges.
- Setting up *routing lists,* each specifying the bridge groups to which calls should be routed in order of preference.

Managing Bridge Groups

This section applies only to routed systems. In a routed system, if you add bridges or change bridge assignments, you must make changes to your bridge groups. By default, your system contains the bridge group All Bridges, which includes all bridges. You can't modify or delete this group. You can add other groups and modify or delete them.

Adding a Bridge Group

1. In the navigation bar, click **Bridge Groups**.

   The **Bridge Groups** page appears (Figure 2-5). It lists existing bridge groups, starting with the default entry, **All Bridges**. A blank input field and **Add** button (green "+") let you add bridge groups. Check boxes below each entry let you select which bridges are in each group.

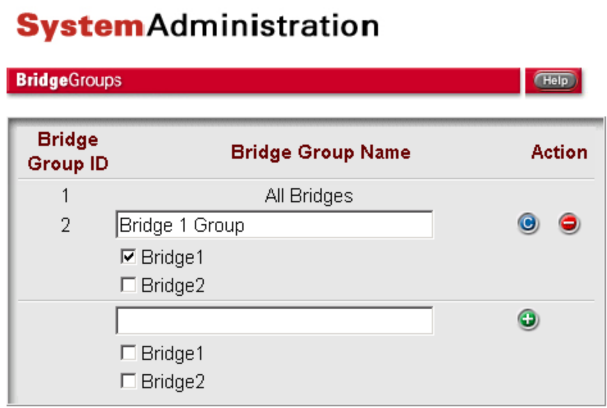
2. Enter a name for the new group in the blank **Bridge Group Name** field.
3. Select the check box of each bridge you want to include in the group and click the **Add** button.

   The system confirms that the bridge group has been added and provides a link back to the **Bridge Groups** page.

4. Click the link to reload the **Bridge Groups** page.

   The new bridge group appears in the list.

Figure 2-5 ***Bridge Groups*** *page (routed system only)*



Repeat the above procedure to add another bridge group. When you're finished, make the necessary changes to the routing lists. See "Managing Routing Lists" on page 27.

Modifying a Bridge Group

1. To change the bridges in the group, select or clear their check boxes.
2. To change the name of the group, edit its **Bridge Group Name** entry.
3. Click the entry's **Commit** button (blue "c").

   The system confirms that the bridge group has been modified and provides a link back to the **Bridge Groups** page.
4. Click the link to reload the **Bridge Groups** page.

Repeat the above procedure to modify another bridge group. When you're finished, make the necessary changes to the routing lists.

Deleting a Bridge Group

1. Click the bridge group's **Delete** button.

   The system confirms that the bridge group has been deleted and provides a link back to the **Bridge Groups** page.
2. Click the link to reload the **Bridge Groups** page.

Repeat the above procedure to delete another bridge group. When you're finished, make the necessary changes to the routing lists.

When you delete a bridge group, the system doesn't make its ID available for reuse or renumber remaining groups. If you have four bridge groups and delete the one whose ID is 3, the remaining bridge groups are 1, 2, and 4. The next bridge group you add is assigned ID number 5. ID number 3 remains unused.

Managing Routing Lists

This section applies only to routed systems (PSTN or IP). In a routed system, routing lists control how calls are routed to the bridges in your system. Each routing list contains one or more bridge groups, listed in order of preference, to which calls can be routed.

By default, your system contains the routing list **Any Bridge**, which contains the default bridge group **All Bridges**. You can't modify or delete this routing list (but need not use it). If you have a routed system, you can add other routing lists and modify or delete them.

Adding a Routing List

1. In the **System Administration** navigation bar, click **Routing Lists**.

   The **Routing Lists** page appears (Figure 2-6). It lists existing routing lists, starting with the default entry, **Any Bridge**. A blank input field and **Add** button (green "+") let you add routing lists.

2. Enter a name for the new routing list in the blank **Routing List Name** field and click the **Add** button.

   The system confirms that the routing list has been added and provides a link back to the **Routing Lists** page.

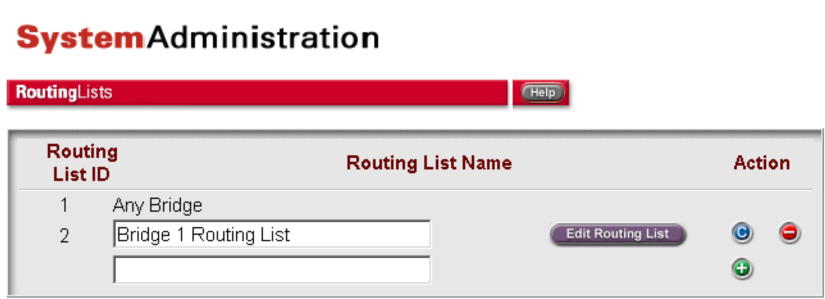
3. Click the link to reload the **Routing Lists** page.

   The new routing list appears in the list. Initially, all new routing lists contain a single bridge group, the default All Bridges group.

4. Modify the routing list to include the bridge groups you want in the order you want them searched (see next procedure).

Repeat the above procedure to add another routing list.

Figure 2-6 ***Routing Lists*** *page (routed system only)*



Modifying a Routing List

1. On the **Routing Lists** page, find the entry you want to modify and click its **Edit Routing List** button.

   The **Edit Routing List** page appears (Figure 2-7). It lists the bridge groups included in this routing list in order of priority. If you just created this routing list, it contains only one bridge group entry, **All Bridges**, with its **Search Order** field set to **1**.

2. To change an existing entry, select a different bridge group from the entry's drop-down list. Then click its **Commit** button (blue "c").

   The system confirms that the routing list has been modified and provides a link back to the **Edit Routing List** page.

   No two entries can have the same priority (**Search Order** value). To change the priority order of the bridge groups, change the **Bridge Group** associated with each **Search Order** value, not vice versa. To avoid confusion, it's best to keep the **Search Order** values of the entries in the list in sequential order.

3. Click the link to reload the **Edit Routing List** page.

4. To add another entry to the list, in the blank row at the bottom, select the bridge group you want, enter the next priority number in the **Search Order** field, and click the **Add** button (green "+").

   The system confirms that the bridge group has been added to the routing list and provides a link back to the **Edit Routing List** page.

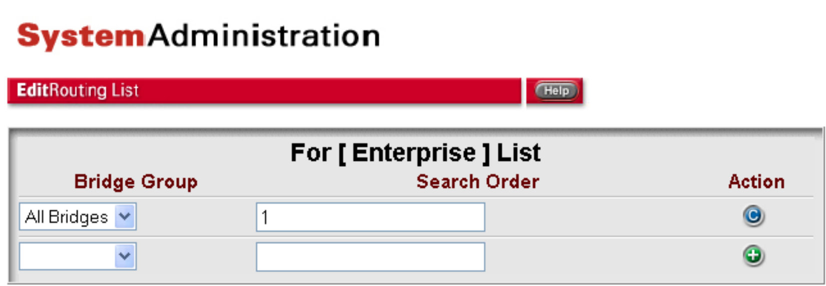
5. Click the link to reload the **Edit Routing List** page.

   The new bridge group appears in the list.

6. To delete an entry from the list, click its **Delete** (red "–") button.

   The system confirms that the bridge group has been removed from the routing list and provides a link back to the **Edit Routing List** page.

Figure 2-7 ***Edit Routing List*** *page (routed system only)*



7 Click the link to reload the **Edit Routing List** page.

The bridge group entry you deleted is gone from the list.

The system doesn't update the **Search Order** numbers of remaining entries. If there were three bridge groups with priority numbers 1, 2, and 3, and you delete the second, the two remaining entries are numbered 1 and 3. Change the entry numbered 3 to 2 in order to maintain sequential numbering and avoid possible confusion in the future.

Repeat the above procedure to modify another routing list.

Renaming a Routing List

1 On the **Routing Lists** page, find the entry you want to rename, edit its **Routing List Name** field, and click its **Commit** button (blue "c").

The system confirms that the routing list has been modified and provides a link back to the **Routing Lists** page.

2 Click the link to reload the **Routing Lists** page.

The renamed routing list entry appears in the list.

Repeat the above procedure to rename another routing list.

Deleting a Routing List

1 On the **Routing Lists** page, find the entry you want to delete and click its **Delete** button (red "–").

The system confirms that the routing list has been deleted and provides a link back to the **Routing Lists** page.

2 Click the link to reload the **Routing Lists** page.

The routing list entry you deleted is gone from the list.

Repeat the above procedure to delete another routing list.

Working with Number Groups

Number groups are logical groupings of telephone numbers used to control the assignment of access phone numbers to subscribers. In a routed system, each number group is associated with a *routing list* that controls the routing of calls among bridges (see "Working with Bridge Groups and Routing Lists" on page 25).

Each number group can have one or more *subscriber groups* associated with it. Subscriber groups are logical groupings of related subscribers who share the same customized user interface (for instance, a specific prompt set). When someone signs up as a new subscriber, the ReadiVoice system automatically assigns the subscriber an access phone number (or numbers) from the number group associated with the subscriber group to which the new subscriber belongs.

You can use a number group to reserve a specific access phone number (or numbers) for a specific subscriber group (or groups). For instance, you may have a subscriber group specifically for Acme Widgets Co. users. If you want all Acme Widgets conferences to use the same access phone number, 888-555-1234, you can:

1. Create a number group called Acme Widgets.
2. Define the access phone number 888-555-1234 as a shared number assigned to the Acme Widgets number group.
3. Assign the Acme Widgets number group to the Acme Widgets subscriber group.

All access phone numbers must be assigned to a number group. One number group, Default, exists automatically. If you don't want to control which access numbers are used by whom, you can assign all numbers to the Default number group and use it for all subscriber groups.

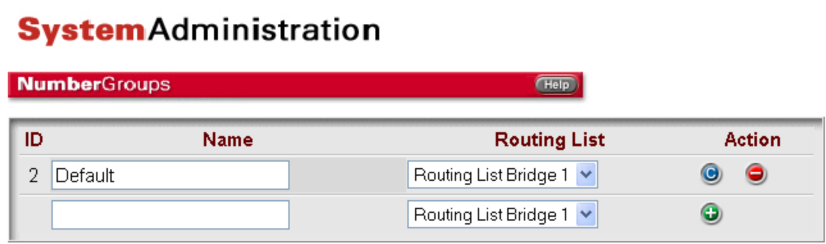
Adding a Number Group

1 In the **System Administration** navigation bar, click **Number Groups**.

   The **Number Groups** page appears (Figure 2-8).The list shows the existing number groups. Below that, a blank input field and **Add** button (green "+") let you add number groups to the list.

2 In the blank **Name** field, enter a name for the number group you want to create. The name may be up to 30 characters long and may contain spaces and punctuation.

Figure 2-8 ***Number Groups*** *page (routed system)*



3 If this is a routed system, choose the routing list to be used for this number group.

4 Click the **Add** button.

   The system confirms that the number group has been added and provides a link back to the **Number Groups** page.

5 Click the link to reload the **Number Groups** page.

   The new group appears in the list.

6 Add one or more access phone numbers to the number group. See "Adding Access Phone Numbers" on page 36.

Repeat the above procedure to create additional number groups. To use a new number group (and the access phone numbers in it), you must assign the number group to one or more subscriber groups. See "Working with Subscriber Groups" on page 46.

Modifying a Number Group

1. In the navigation bar, click **Number Groups**.

   The **Number Groups** page appears, listing the available number groups.

2. To rename a number group, edit its name.
3. To change its routing list assignment, select the one you want in the **Routing List** field.
4. Click the **Commit** button (blue “c”).

   The system confirms that the number group has been modified and provides a link back to the **Number Groups** page.

5. Click the link to reload the **Number Groups** page.

Repeat the above procedure to modify another number group.

Deleting a Number Group

You can delete a number group only if it has no associated subscriber group(s) and no associated access phone numbers.

To delete a number group:

1. In the navigation bar, click **Number Groups**.

   The **Number Groups** page appears, listing the available number groups.

2. Click the **Delete** button (red “–”) by the number group you want to remove.

   The system confirms that the number group has been deleted.

   When you delete a number group, the system doesn’t make its ID available for reuse or renumber remaining groups. If you have four number groups and delete the one whose ID is 3, the remaining groups are 1, 2, and 4. The next number group you add is assigned ID number 5. ID number 3 remains unused.

Repeat the above procedure to delete another number group.

Working with Access Classes

Access classes are logical groupings of access phone numbers you can use to group numbers by carrier, kind of number (such as toll-free), or other criteria. You can create as many classes as you want. A subscriber can have one access phone number from each access class, so multiple classes let you offer subscribers multiple access numbers.

All access phone numbers belong to an access class. One class, LOCAL, exists automatically. You can rename it if you want. If you don't want to categorize access numbers by carrier or other criteria and you don't want to offer multiple access numbers to subscribers, you can assign all numbers to one class.

Adding an Access Class

1 In the **System Administration** navigation bar, click **Access Classes**.

   The **Access Classes** page appears (Figure 2-9). The list shows the existing access classes. Below that, a blank input field and **Add** button (green "+") let you add access classes to the list.

2 In the blank **Name** field, enter a name for the access class. It may be up to 30 characters long and may contain spaces and punctuation.

3 Click the **Add** button.

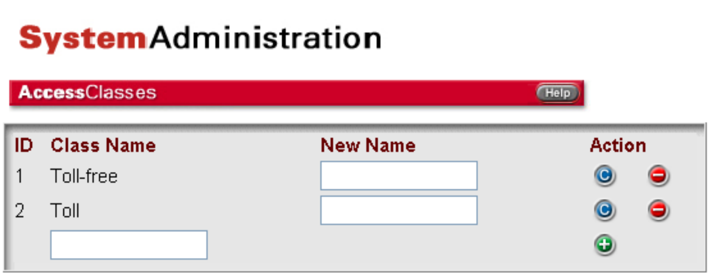
   The system confirms that the access class has been added and provides a link back to the **Access Classes** page.

4 Click the link to reload the **Access Classes** page.

   The new access class appears in the list.

   Repeat the above procedure to create more access classes. Then, assign access phone numbers to the classes. See "Working with Access Phone Numbers" on page 35.

Figure 2-9 ***Access Classes** page*



Renaming an Access Class

1. In the navigation bar, click **Access Classes**.

   The **Access Classes** page appears, listing the available access classes.

2. Find the class you want to rename and type the new name into its **New Name** field.

3. Click its **Commit** button (blue “c”).

   The system confirms that the access class has been modified and provides a link back to the **Access Classes** page.

4. Click the link to reload the **Access Classes** page.

   The new class appears in the list.

Repeat the above procedure to rename another access class.

Deleting an Access Class

You can’t delete an access class if it contains access phone numbers. First, reassign those numbers to another class. You can do this only if the access numbers have no subscribers associated with them. See “Changing an Access Phone Number’s Access Class” on page 40.

To delete an access class:

1. In the navigation bar, click **Access Classes**.

   The **Access Classes** page appears, listing the available access classes.

2. Click the **Delete** button (red “–”) by the access class you want to remove.

   If the class has access numbers in it, the system displays an error message. Otherwise, it confirms that the access class has been deleted and provides a link back to the **Access Classes** page.

3. Click the link to reload the **Access Classes** page.

   When you delete an access class, the system doesn’t make its ID available for reuse or renumber the remaining classes. If you have four access classes and delete the one whose ID is 3, the remaining classes are 1, 2, and 4. The next access class you add is assigned ID number 5. ID number 3 remains unused.

Repeat the above procedure to delete another access class.

Working with Access Phone Numbers

Access phone numbers are the telephone numbers (or, for IP systems, SIP addresses) that customers dial to reach your ReadiVoice system. They're often toll-free numbers, and your carrier usually tells you which numbers to use. In the ReadiVoice system, access numbers have several characteristics:

Access Type. A routed system can have two access types:

> *Private* — A private number is assigned to only one subscriber. Since the access phone number uniquely identifies a subscriber's conference, no access code is needed.
>
> *Shared* — A shared number is assigned to multiple subscribers. Each subscriber receives a unique access code that identifies the subscriber's conferences. Callers must enter the correct access code to join a conference.
>
> For a non-routed PSTN system, all access phone numbers must be either private or shared; you can't mix the two. For an IP system, access numbers are shared.

Number Group. Each access phone number belongs to one number group, which can be associated with one or more subscriber groups. The access phone number is available in subscriber groups that use its number group.

Access Class. Each access phone number belongs to an access class. Multiple classes let you offer subscribers multiple access numbers, one from each class. You can create classes for specific kinds of numbers, such as local or toll-free.

Hidden Number (if enabled; see "Changing System Configuration Settings" on page 118). If the network maps the numbers that callers dial to different numbers for routing, the system must link the routing number (hidden number) with the dialed number.

You can:

- Add access phone numbers.
- In a non-routed system, move access numbers from one bridge to another.
- Globally reassign subscribers from one access number to another.
- Change the access class of an access phone number (if no subscribers are using it).
- Change the hidden number associated with an access phone number (if hidden number mapping is enabled).
- Delete an access phone number (if no subscribers are using it).

You can't change the access type or number group of an existing access number. To change a number from private to shared or switch it to a different number group, delete the number and then add it back.

Adding Access Phone Numbers

To put the new access phone numbers into a new number group or access class, create the number group or access class first. See "Adding a Number Group" on page 31 or "Adding an Access Class" on page 33.

To add an access phone number:

1 In the **System Administration** navigation bar, click **Access Numbers**.

The **Access Phone Numbers** page appears (Figure 2-10). The **Search** panel is at the top. Below it is the **Add** panel.

2 In the **Add** panel, enter the access number you want to add. Enter only the digits, with no dashes or spaces.

3 If your system uses hidden numbers, enter the new number's hidden number. Enter only the digits, with no dashes or spaces.

4 Set the new access number's **Number Group** and **Access Class** fields.

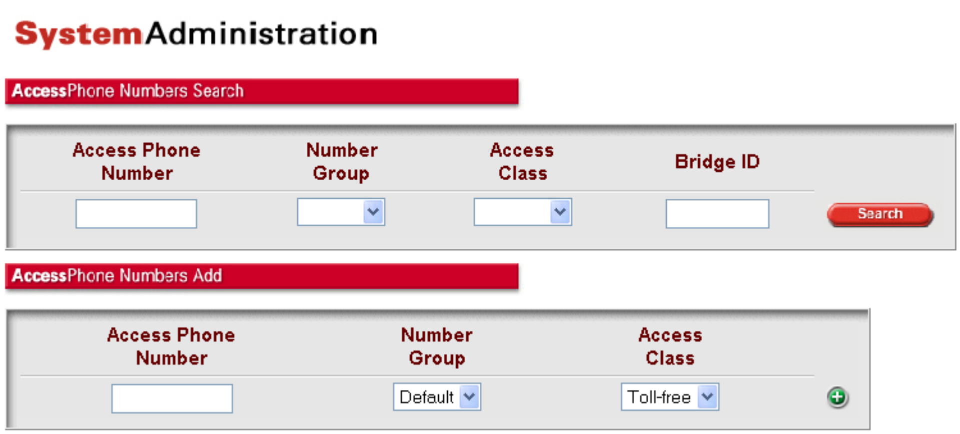
5 Click the **Add** button (green "+").

The system confirms that the access phone number has been added and provides a link back to the **Access Phone Numbers** page.

6 Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to add another access phone number.

***Figure 2-10 Access Phone Numbers** page*



Moving Fixed Access Numbers to Another Bridge

In non-routed systems, access numbers are assigned to specific bridges. If necessary, you can move access numbers from one bridge to another. For example, you may want to move some access numbers from an overloaded bridge to another bridge that's less loaded.

To move access numbers to a different bridge:

1. In the navigation bar, click **Access Numbers**.

   The **Access Phone Numbers** page appears (Figure 2-10).

2. In the **Search** panel, enter criteria to retrieve the access numbers you want to move. Then click the **Search** button.

   You can use the asterisk (*****) as a wild card. For instance, enter **80033311*** to retrieve all numbers from 800-333-1100 to 800-333-1199.

   The access numbers that match your search criteria appear at the bottom of the page in the **Numbers** panel (Figure 2-11).

3. Carefully check the list of access numbers retrieved. If the list contains the numbers you want to move and *only* those numbers, continue. Otherwise, repeat 2 using different search criteria until you've retrieved the correct set of numbers.

4. In the **Move** panel, select the bridge to which you want to move the access numbers. Then click the **Move** button.

   The system prompts you to confirm the move and reminds you to make the telephone network changes needed to implement the move.

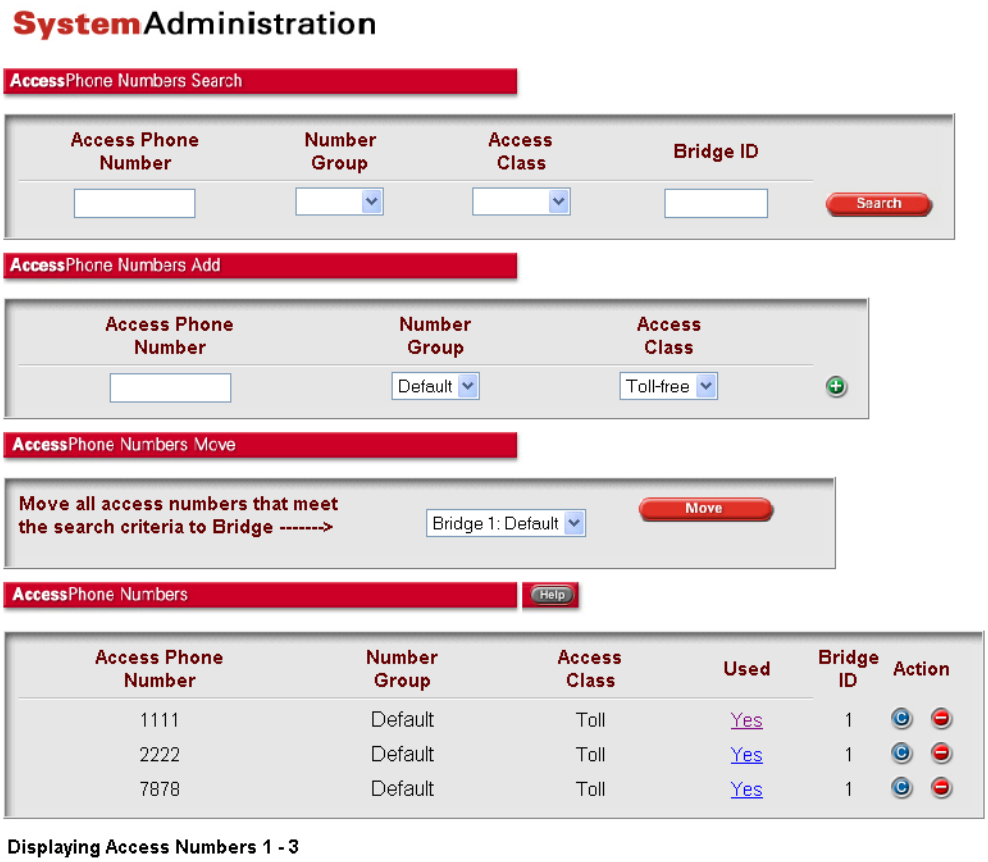
5. Click **OK**.

   The system confirms that the access numbers have been moved and provides a link back to the **Access Phone Numbers** page.

6. Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to move additional blocks of access numbers. When you're finished changing bridge assignments, make the necessary changes to switches or other network elements in order to correctly route calls dialed to these access numbers.

Figure 2-11 ***Access Phone Numbers*** *page after search*



Reassigning an Access Phone Number's Subscribers

You can globally reassign subscribers from one access number to another. For instance, if several subscribers are assigned to the shared access number 888-555-1000, and you need to reassign them to 888-555-2000, you can make this change all at once without having to re-provision each individual subscriber.

You can:

- Reassign subscribers to a different *existing* access phone number. The number must be in the same number group and access class.
- Reassign subscribers to a *new* access phone number, adding it to the ReadiVoice system in the process.

You must, of course, notify the reassigned subscribers of the change. If you add a new access phone number to the system, you must also ensure that the telephone network routes that number to the system properly.

To reassign subscribers to a different access phone number:

1. In the navigation bar, click **Access Numbers**.

   The **Access Phone Numbers** page appears (Figure 2-10).

2. In the **Search** panel, enter criteria to retrieve the access number whose subscribers you want to reassign. Then click the **Search** button.

   The access numbers that match your search criteria appear at the bottom of the page in the **Numbers** panel (Figure 2-11).

3. In the **Numbers** panel, find the access number whose subscribers you want to reassign and click its **Change** button (blue "c").

   The **Modify Existing Access Phone Number** page appears (Figure 2-12).

4. In the **New Access Phone Number** field, enter the number you want to assign to the subscribers. Enter only the digits, with no dashes or spaces. If this is an existing access number, it must be in the same number group and access class as the access number it's replacing.

5. If you're creating a new access phone number for these subscribers and your system uses hidden numbers, enter the access number's hidden number in the **New Hidden Phone Number** field.

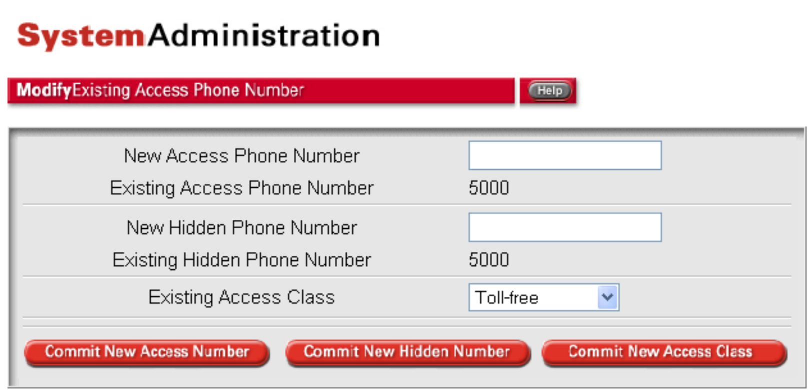
6. Click **Commit New Access Number**.

   If you entered a new number, the ReadiVoice system adds it to the database, copying the access type, number group, and access class from the existing number. It confirms that all the existing number's subscribers have been reassigned to the access phone number you entered and provides a link back to the **Access Phone Numbers** page.

7. Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to reassign another access phone number's subscribers.

***Figure 2-12 Modify Existing Access Phone Number** page*



Changing an Access Phone Number's Access Class

You can't change an access phone number's access class if subscribers are assigned to it. First, reassign the subscribers to another access number. See "Reassigning an Access Phone Number's Subscribers" on page 38.

To change an access number's access class:

1. In the navigation bar, click **Access Numbers**.

   The **Access Phone Numbers** page appears (Figure 2-10).

2. In the **Search** panel, enter criteria to retrieve the access number whose access class you want to change. Then click the **Search** button.

   The access numbers that match your search criteria appear at the bottom of the page in the **Numbers** panel (Figure 2-11).

3. Confirm that the access number you want to change is listed and has no subscribers assigned to it (**Used** is set to **No**).

   If the **Used** field is set to **Yes**, click that word to see a list of the number's subscribers (Figure 2-13). You can reassign them with the procedure on page 40 or, if there are only one or two, open and edit them manually.

4. Click the **Change** button (blue "c") of the number you want to change.

   The **Modify Existing Access Phone Number** page appears (Figure 2-12).

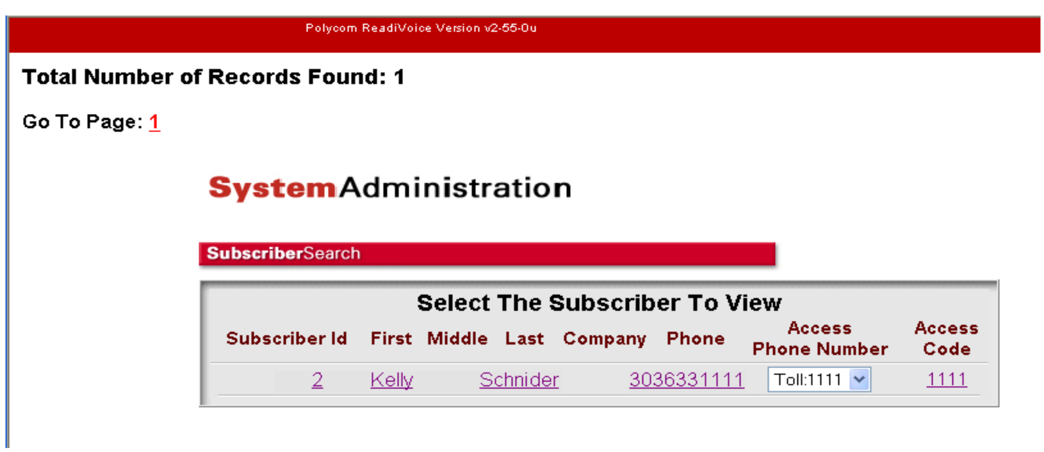
5. In the **Existing Access Class** list, select the class to which you want to assign this access number. Then click **Commit New Access Class**.

   The system confirms that the access class has been changed and provides a link back to the **Access Phone Numbers** page.

6. Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to change another access phone number.

Figure 2-13 Checking an Access Phone Number's subscribers



Changing Hidden Numbers

This section doesn't apply to IP systems, which don't use hidden numbers.

If your system uses hidden numbers, you can use the **Modify Existing Access Phone Number** page to change the hidden number associated with an access number. Do this *only* if the routing number associated with the access number is changed in the network.

To change an access phone number's hidden number:

1. In the navigation bar, click **Access Numbers**.

   The **Access Phone Numbers** page appears (Figure 2-10).

2. In the **Search** panel, enter criteria to retrieve the access number whose hidden number you want to change. Then click the **Search** button.

   The access numbers that match your search criteria appear at the bottom of the page in the **Numbers** panel (Figure 2-11).

3. Find the access number whose hidden number you want to change and click its **Change** button (blue "c").

   The **Modify Existing Access Phone Number** page appears (Figure 2-12).

4. In the **New Hidden Phone Number** field, enter the new hidden number to associate with this access phone number. Enter only the digits, with no dashes or spaces. The hidden number must be unique (that is, not already assigned to another access phone number).

5. Click **Commit New Hidden Number**.

   The system confirms that the hidden number has been changed and provides a link back to the **Access Phone Numbers** page.

6. Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to change another access phone number's hidden number.

Deleting an Access Phone Number

Before you delete an access phone number, keep the following points in mind:

- Access phone number changes require telephone network changes to ensure that calls are routed properly.
- You can’t delete an access phone number that has subscribers assigned to it. First, reassign the subscribers to another access number. See “Reassigning an Access Phone Number’s Subscribers” on page 38.

To delete an access phone number:

1. In the navigation bar, click **Access Numbers**.

   The **Access Phone Numbers** page appears (Figure 2-10 on page 36).

2. In the **Search** panel, enter criteria to retrieve the access number that you want to delete. Then click the **Search** button.

   The access numbers that match your search criteria appear at the bottom of the page in the **Numbers** panel (Figure 2-11).

3. Confirm that the access number you want to delete is listed and has no subscribers assigned to it (**Used** is set to **No**).

   If the **Used** field is set to **Yes**, click that word to see a list of the number’s subscribers (Figure 2-13). You can reassign them with the procedure on page 40 or, if there are only one or two, open and edit them manually.

4. Click the **Delete** button (red “–”) of the number you want to delete.

   The system confirms that the number has been deleted and provides a link back to the **Access Phone Numbers** page.

5. Click the link to reload the **Access Phone Numbers** page.

Repeat the above procedure to delete another access phone number.

Working with Multiple Providers

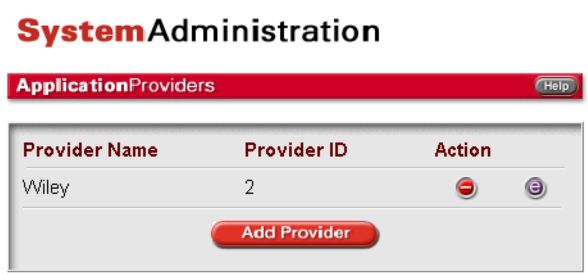
You can configure ReadiVoice for access by multiple *application service providers* (ASPs). For instance, your company can enter into a relationship with Acme Conferencing, allowing Acme to sell services on your ReadiVoice system, using their own custom interfaces (such as the Moderator interface and voice prompts). To enable such a relationship, you configure Acme as a provider in your ReadiVoice system, permitting their users to connect to it.

Adding a Provider

1 In the navigation bar, click **Application Providers**.

The **Application Providers** page appears (Figure 2-14), listing any providers already defined in the system.

***Figure 2-14 Application Providers** page*



2 Click **Add Provider**.

The **Add Provider** page appears (Figure 2-15 on page 45).

3 Enter the data for the provider. See Table 2-2 for field descriptions.

4 Click **Create Provider**.

The system confirms that the provider has been added and provides a link back to the **Application Providers** page.

5 Click the link to reload the **Application Providers** page.

Repeat the above procedure to add another provider.

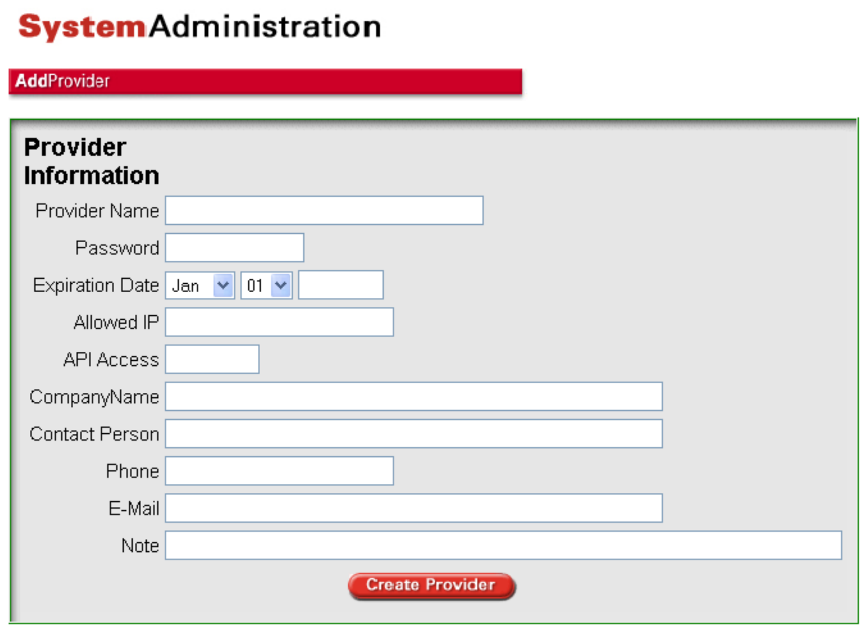
***Table 2-2** Provider Information fields*

Field	Description
Provider Name	Enter a login name for the provider. The maximum length is 30 characters. Required.
Password	Enter a password for the provider. The maximum length is 30 characters. Required.
Expiration Date	Select the expiration month and date and enter a four-digit year. Required.

Table 2-2 Provider Information fields (continued)

Field	Description
Allowed IP	Enter a specific IP or an IP mask that defines valid IP addresses from which this provider can connect. For instance, if you enter 192.168.56.255, all IP addresses in the 192.168.56 subnet are valid.
API Access	Define which APIs the provider can access by entering the sum of the authorized APIs' values. Required. The API Values are: **1** Moderator API **2** Operator API **4** Call Router API **8** Provisioning API **16** CDR API For instance, to authorize access to the Moderator and Operator APIs only, enter 3. To authorize access to the Moderator, Operator, and Provisioning APIs, enter 11. Currently, only the Moderator API can be used.
Company Name	Enter the provider's company name. The maximum length is 30 characters.
Contact Person	Enter the name of the person responsible for this account. The maximum length is 30 characters.
Phone	Enter a phone number for the contact person. The maximum length is 30 characters.
E-Mail	Enter an email address for the contact person. The maximum length is 50 characters.
Note	Enter any additional information about the provider that you need. The maximum length is 256 characters.

*Figure 2-15 **Add Provider** page*



Editing a Provider Account

1. In the navigation bar, click **Application Providers**.

   The **Application Providers** page appears (Figure 2-14).

2. Find the provider that you want to edit and click its **Edit** button (purple "e").

   The **Add Provider** page appears (Figure 2-15).

3. Modify the data as necessary. See Table 2-2 for field descriptions.

4. Click **Create Provider**.

   The system confirms that the provider has been modified and provides a link back to the **Application Providers** page.

5. Click the link to reload the **Application Providers** page.

Repeat the above procedure to modify another provider.

Deleting a Provider

You can't delete a provider account to which subscribers are assigned.

To delete a provider account from the system:

1. In the navigation bar, click **Application Providers**.

   The **Application Providers** page appears (Figure 2-14).

2. Find the provider you want to delete and click its **Delete** button (red "–").

   A dialog box asks you to confirm.

3. Click **OK**.

   The system confirms that the provider has been deleted and provides a link back to the **Application Providers** page.

4. Click the link to reload the **Application Providers** page.

Repeat the above procedure to delete another provider.

Working with Subscriber Groups

Subscriber groups are logical groupings of related subscribers who share the same user interface (for instance, a specific prompt set). All subscribers are in a group, so you must have at least one. Typically, you create more subscriber groups as needed. For instance, you may create a subscriber group specifically for Acme Widgets Co. users.

During ReadiVoice installation, a default subscriber group is automatically created. Initial configuration of the system includes defining the initial value settings for the default subscriber group.

A subscriber group's settings are used as initial values for its new subscribers. Therefore, the settings you make for a group are the defaults that provisioners see when they add subscribers to the group.

After working with the subscriber group settings, you may realize that your company doesn't need or want to use all the available fields when provisioning subscribers or modifying their accounts. See "Setting Up the Provisioning Interface" on page 60.

When you create a new group, the system copies the settings from the existing group of your choice to save you time.

Each subscriber group can have a *group administrator*, who is the internal user authorized to modify the group and view its CDR and conference information. You can't directly create or delete group administrators as you do other internal users. You can do so only as part of creating or editing the group. You can, however, change the group administrator's password directly on the Passwords page. See "Changing an Internal User's Password" on page 65.

The following sections describe how to create, edit, and delete a group. For information about adding subscribers to a group, see the *ReadiVoice Provisioning Guide*.

Adding a New Subscriber Group

Every subscriber group must be associated with a number group, which controls the access numbers available to the subscriber group. If you want to link the new subscriber group with its own number group, create the number group first. See "Working with Number Groups" on page 30.

To add a new subscriber group:

1 In the **System Administration** navigation bar, click **Subscriber Groups**.

The **Subscriber Group** page appears (Figure 2-16). The **List** panel shows the subscriber groups that exist.

2 In the **Add** panel, select an existing group from which to copy the initial values and enter a name for the new group. The maximum length is 50 characters.

3 Click the **Add New Group** button.

The system creates the new group, copying its initial values from the existing group you selected, and opens the **Edit Group** page (Figure 2-17).

4 Edit the group information as described in "Editing a Subscriber Group," starting at 3.

Repeat the above procedure to add another subscriber group.

Figure 2-16 Subscriber Group page

Editing a Subscriber Group

1 In the navigation bar, click **Subscriber Groups**.

The **Subscriber Group** page appears.

2 Find the group that you want to edit and click its **Edit Group** button (purple “e”).

The **Edit Group** page appears (Figure 2-17).

3 Enter or modify the data in the **Group Information** area. See Table 2-3 for descriptions.

***Figure 2-17 Edit Group** page (top portion)*

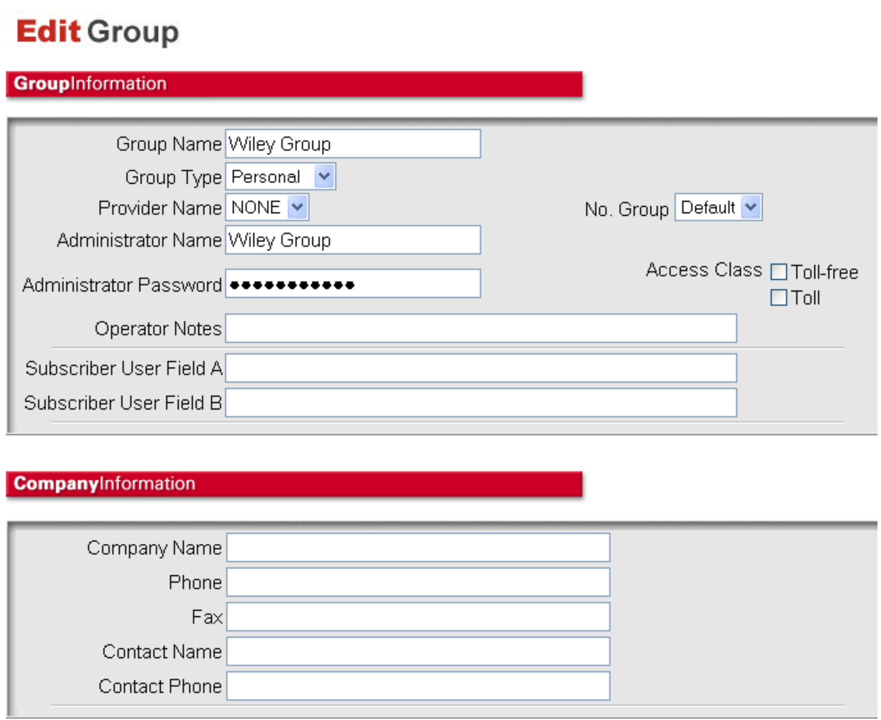


Table 2-3 *Group Information fields*

Field[a]	Description
Group Name	Leave the current name or change it if you wish. The maximum length is 50 characters.
Group Type	Choose Personal or Corporate from the list. Personal is for billing each participant individually. Corporate is for billing a company for the entire conference. If you choose Corporate, you must complete the Company Information and Billing Information sections.
Provider Name	If your system has multiple providers, select the provider for this subscriber group from the list. Otherwise, leave the default value of **NONE**.
Number Group	Select the number group from which access phone numbers for this group are assigned.
Administrator Name	Change the login name of the group administrator (the initial value is the group name). This is the person authorized to change group information and view CDR (call detail record) and conference information for this group. The maximum length is 30 characters.
Administrator Password	Change the password for this group administrator (the initial value is the group name). The maximum length is 30 characters.
Access Class	Place a check next to the access class or classes that should be selected by default for new subscribers in this group. A class typically defines a type of access number, such as toll or toll-free.
Operator Notes	Enter any message or operator script that you want displayed to operators when they view this group's conferences. The maximum length is 484 characters.
Subscriber User Field A	Optional field for additional subscriber group information, such as a key to another database. Whether and how you use this field is entirely up to your company. The ReadiVoice system doesn't use it.
Subscriber User Field B	Second optional field for additional subscriber group information.

a. These are the default labels. All field labels that also appear on the New Subscriber page can be customized.

4 If **Group Type** is Corporate:

a Enter or modify the data in the **Company Information** fields. See Table 2-4 for descriptions. All fields are required.

b Enter or modify the data in the **Billing Information** section. See Table 2-5 for descriptions.

Figure 2-18 shows these two sections.

The ReadiVoice system doesn't use any of the data in the **Billing Information** section. These fields are provided for your company to use as it wishes, and the descriptions here are just general guidelines.

Figure 2-18 ***Company*** *and* ***Billing*** *sections of* ***Edit Group*** *page*

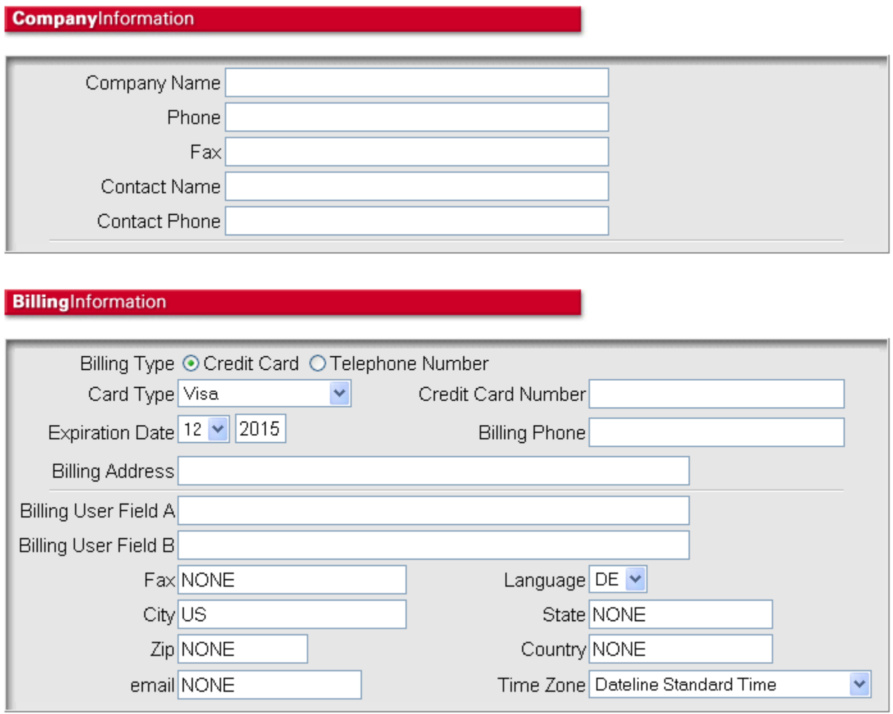


Table 2-4 *Company Information fields*

Field	Description
Company Name	Enter the name of the company. The maximum length is 50 characters.
Phone	Enter the company’s telephone number. The maximum length is 30 characters.
Fax	Enter the company’s fax number. The maximum length is 30 characters.
Contact Name	Enter the name of the person to contact at this company. The maximum length is 50 characters.
Contact Phone	Enter the contact person’s telephone number. The maximum length is 30 characters.

Table 2-5 *Billing Information fields*

Field[a]	Description
Billing Type	Select Credit Card or Telephone Number to choose the billing method for this group's conferences.
Card Type	For credit card billing, select the card type from the list.
Credit Card Number	For credit card billing, enter the 16-digit card number.
Expiration Date	For credit card billing, select the expiration month and enter the four-digit year.
Billing Phone	Enter the phone number to be billed or the phone number from the Company Information section. This field is required, even if Billing Type is not Telephone Number.
Billing Address	Enter the billing address.
Billing User Field A	Optional field for additional billing information, such as a key to another database. Whether and how you use this field is entirely up to your company.
Billing User Field B	Second optional field for additional billing information.

a. These are the default labels. All field labels that also appear on the New Subscriber page can be customized.

5 Configure the prompt settings in the **Call Flow and Voice Prompt** section (Figure 2-19). Table 2-6 describes the settings.

Figure 2-19 ***Call Flow and Voice Prompt*** *section of* ***Edit Group*** *page*

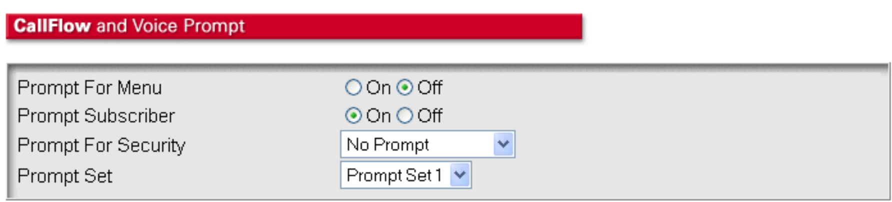


Table 2-6 *Call Flow and Voice Prompt fields and settings*

Field/Setting[a]	Description
Prompt for Menu	Specifies whether subscribers can access the account options menu when they dial into the system and identify themselves. **On** — Subscribers in this group are given the opportunity to change the account options. The account options menu enables subscribers to make changes that affect all future conferences (including the one that a subscriber is in the process of starting). **Off** — Subscribers are not given the opportunity to access the account options menu (but can still modify account options through the Moderator interface).
Prompt for Subscriber	Indicates whether the call flow includes a prompt for the subscriber to enter his or her subscriber password. **On** — Callers hear "*If you are the subscriber, press star*" (this is the default wording). Use this setting *unless* all subscribers in this group *must* have a two-password (shared or private) call flow. **Off** — Callers hear "*At any time during this message, please enter your passcode followed by the pound key.*" Use this setting if *only* two-password call flows are possible. If this is set to Off for a subscriber account with a traditional (one password) call flow, the subscriber has no opportunity to identify him or herself when dialing in and must use the Moderator to start conferences.
Prompt for Security	Specifies the setting for the conference security code prompt. **No Prompt** — The conference security feature isn't available. **Optional Prompt** — Subscribers are given the option of setting up a conference security code, which participants must know in order to join the conference. **Mandatory Prompt** — Subscribers are required to set up a conference security passcode.
Prompt Set	Specifies the voice prompt set for this subscriber group's conferences.

a. These are the default labels. All field labels that also appear on the New Subscriber page can be customized.

6 In the **Conference Options** section (Figure 2-20), configure Account Options and Roll Call Options. Each of the features has two settings:

Setting — Determines the feature's default setting.

Subscriber Configurable — Determines whether subscribers can change their account's default setting later, using either the account options call flow menu (if enabled) or the Moderator.

Table 2-7 on page 54 describes the settings. Remember, the group's settings are the initial settings for subscribers in the group, but provisioners can modify them for specific subscribers.

7 Continue with the **Conference Options** section by setting the Provisioning Options. See Table 2-7.

8 If your system has any Application Control Mode (ACM) features installed, place a check mark next to those that you wish to enable for this subscriber group.

ACM applications must be both installed and activated to appear here.

Figure 2-20 Conference Options *section of* ***Edit Group*** *page*

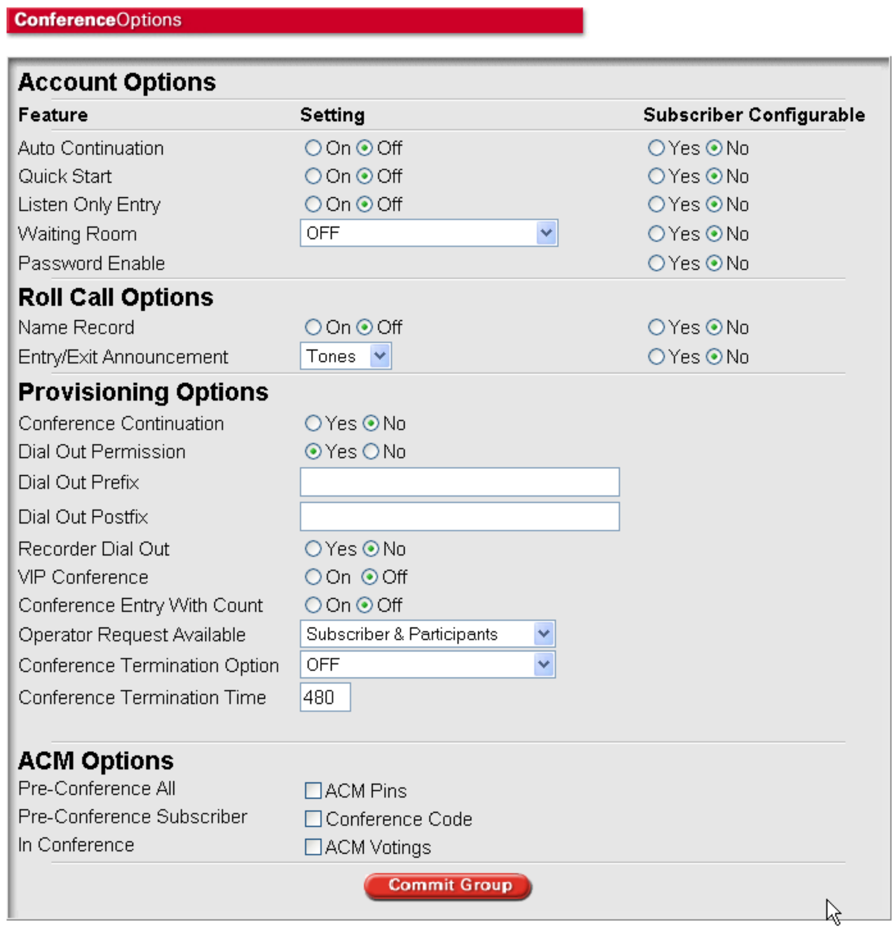


Table 2-7 *Conference Options*

Field[a]	Description
Auto Continuation	**Setting:** **On** — By default, conferences begin with continuation turned on (which means that the conference continues until the last participant disconnects). **Off** — By default, conferences begin with continuation turned off (which means that the conference ends when the subscriber disconnects). **Subscriber Configurable:** **Yes** — Subscribers can change the On/Off setting *before* a conference begins through the account options menu (if available) or the Moderator interface. **No** — Subscribers can't change the On/Off setting. The **Conference Continuation** setting determines whether subscribers can *override* the default continuation behavior for a specific conference. If you set **Subscriber Configurable** to **No** and **Conference Continuation** to **No**, then subscribers can neither *change* the default behavior nor *override* it; the **Auto Continuation Setting** (**On** or **Off**) governs all of a subscriber's conferences. See "How Continuation Settings Interact" on page 58.
Quick Start	**Setting:** **On** — By default, conferences start as soon as the first participant dials in. Participants don't wait on hold for the subscriber to arrive. **Off** — By default, conferences start when the subscriber arrives. Participants who dial in before the subscriber wait on music hold until the subscriber arrives. **Subscriber Configurable:** **Yes** — Subscribers can change the On/Off setting *before* a conference begins through the account options menu (if available) or the Moderator interface. **No** — Subscribers can't change the On/Off setting. Quick start conferences are considerably less secure than conferences that don't start until the subscriber enters the subscriber password.
Listen Only Entry	**Setting**: **On** — Conferences start in listen only mode. The subscriber is the only conference participant who can speak to the conference; all other participant lines are muted, and participants can't unmute themselves. **Off** — Conferences don't start in listen only mode. All participants (except those who are muted) can be heard. **Subscriber Configurable**: **Yes** — Subscribers can change the On/Off setting for the account, affecting all future conferences. **No** — Subscribers *can't* change the On/Off setting. But, they can still use the Moderator interface or touchtone commands to turn listen only mode on or off in the current conference.

Table 2-7 Conference Options (continued)

Field[a]	**Description**
Waiting Room	**Setting**: **Off** — The waiting room isn't available. If a subscriber locks a conference, callers after that are told that the conference is locked and are disconnected. **On** — The waiting room is available. If a subscriber locks a conference, callers after that must wait on music hold to be admitted. The subscriber is notified when someone wants to enter the conference and can use touchtone commands or the Moderator interface to speak with, admit, or disconnect the waiting caller(s). **On w/Waiting Room on Entry** — Conferences start locked with the waiting room available. All callers must wait to be admitted by the subscriber. **Subscriber Configurable**: **Yes** — Subscribers can change the default setting for their account, affecting all future conferences. **No** — Subscribers *can't* change the default. If a subscriber locks the current conference, the waiting room either is or isn't available, depending on the setting for this feature.
Password Enable	Determines whether subscribers can change their subscriber password using the touchtone interface.
Name Record	**Setting**: **On** — The ReadiVoice system prompts participants to record their names before placing them into conference. Conferees can request a "roll call" of participants' names during the conference. This feature must be turned on if **Entry/Exit Announcement** is set to **Name**. **Off** — The ReadiVoice system doesn't record participants' names. The **Entry/Exit Announcement** setting can't be **Name**. **Subscriber Configurable**: **Yes** — Subscribers can change the On/Off setting for the account, affecting all future conferences. **No** — Subscribers *can't* change the On/Off setting.
Entry/Exit Announcement	**Setting**: Specifies whether the system signals the entry and exit of a participant by playing a tone, the participant's recorded name, or silence. If you select **Name**, you must also turn on the **Name Record** feature. **Subscriber Configurable**: **Yes** — Subscribers can change the default setting for the account, affecting all future conferences. If the announcement setting is **Name**, and if the subscriber can turn off **Name Record**, then the subscriber should be able to change the announcement setting, too. **No** — Subscribers *can't* change the default.

Table 2-7 *Conference Options (continued)*

Field[a]	Description
Conference Continuation	Determines whether subscribers can *override* their accounts' default continuation behavior (specified by the **Auto Continuation** settings) for a specific conference *during* that conference (using either a touchtone command or the Moderator interface). **Yes** — Subscribers can turn continuation on or off during their conferences. Doing so doesn't change the default (**Auto Continuation**) setting for the account; only the current conference is affected. **No** — Subscribers can't override the default continuation behavior during their conferences. Once a conference has started, the account's **Auto Continuation** setting controls the continuation behavior. If **Conference Continuation** is **No** and **Auto Continuation** isn't subscriber configurable, then the subscriber can neither *change* the default behavior nor *override* it; the **Auto Continuation Setting** (**On** or **Off**) governs all the subscriber's conferences. See "How Continuation Settings Interact" on page 58.
Dial Out Permission	Determines whether a subscriber can dial out from his or her conferences.
Dial Out Prefix	Appears only if your system is configured for dial-out billing (see "Configuring Dial-Out Billing" on page 126). Enables billing for dial-out calls a subscriber makes. This field and **Dial Out Postfix** together can't exceed 24 characters. This field has no effect if **Dial Out Permission** is set to **No**.
Dial Out Postfix	Appears only if your system is configured for dial-out billing. Enables billing for dial-out calls the subscriber makes. This field and **Dial Out Prefix** together can't exceed 24 characters. This field has no effect if **Dial Out Permission** is set to **No**.
Recorder Dial Out	Determines whether a subscriber can choose to record a conference. Conference recording is available only if your system is configured properly and can dial out to a recording service or device. See "Enabling Conference Recording" on page 137 and ask your Polycom Global Services representative about conference recording.
VIP Conference	**On** — Subscribers in this group have VIP Conference turned on by default. VIP Conference subscribers are *guaranteed* access to the number of ports specified in their subscriptions. Polycom recommends selecting **Off** for this option. It's best to grant VIP status only to individual subscribers and only in limited numbers. VIP subscribers have exclusive use of their ports at all times, permanently removing those ports from the pool of available ports for non-VIP conferences. Overuse of the VIP Conference option has a negative impact on your system's port utilization efficiency. **Off** — Subscribers in this group have VIP Conference turned off by default. This is the normal setting. Non-VIP subscribers' ports come from the general pool, and the full number specified in their subscriptions is not guaranteed to be available at all times.

Table 2-7 *Conference Options (continued)*

Field[a]	**Description**
Conference Entry with Count	**On** — Upon entering the conference, participants hear a private message telling them how many people, including themselves, are now in the conference. **Off** — Participants hear a message stating only that they're being placed into conference.
Operator Request Available	Specifies who may make operator requests from a subscriber's conferences: the subscriber only, the subscriber and participants, or neither. This setting doesn't affect automatic operator requests resulting from a caller's failure to enter the correct password or conference security code.
Conference Termination Option	Specifies how the ReadiVoice system handles conferences that have the minimum number of participants: **Off** — The ReadiVoice system never terminates conferences. **On** — After a configurable interval (set by **Conference Termination Time**, below), the conference is referred to an operator for termination. If no operator is available, the system plays a message to the conference asking the participants to press a touchtone key to continue. If there's no response, the system terminates the conference. **On – w/ Oper Override** — The system plays a message to the conference asking the participants to press a touchtone key to continue. If there's no response, the system terminates the conference. You can set the minimum number of participants for your system to one, two, or three. See "Changing System Configuration Settings" on page 118.
Conference Termination Time	Specifies the time interval (in minutes) that governs the automatic termination of conferences that have the minimum number of participants. Has no effect if **Conference Termination Option** is set to **Off**.
ACM Options	Application Control Mode (ACM) enables an external application to interact with callers or conference participants. The options or settings that appear here, if any, depend on which ACM features and external applications are installed in your system. ACM and some sample ACM applications to demonstrate its use are included in the ReadiVoice Software Development Kit (SDK).

a. These are the default labels. All field labels that also appear on the New Subscriber page can be customized via the Quick Provisioning Setup page described in "Setting Up the Provisioning Interface" on page 60.

9 Click **Commit Group**.

The system either confirms that the group has been updated or tells you which information is missing.

10 If the update failed, note what's missing. Then click the link to return to the **Edit Group** page, make the needed corrections, and click **Commit Group** again.

Repeat the above procedure to edit another subscriber group.

How Continuation Settings Interact

Table 2-8 shows how **Auto Continuation** and **Conference Continuation** interact to determine an account's continuation behavior (whether a conference continues after the subscriber leaves) and what control the subscriber has over that behavior.

Table 2-8 *Continuation settings at a glance*

Auto Continuation			
Setting	**Subscriber Configurable**	**Conference Continuation**	**Effect**
On	**Yes**	**Yes**	Continuation on. Subscriber can turn it off for the current conference or for all future conferences.
		No	Continuation on. Subscriber can turn it off for all future conferences, but not for the conference currently under way.
	No	**Yes**	Continuation on. Subscriber can turn it off for the conference currently under way, but not for all future conferences.
		No	Continuation on. Subscriber can't turn it off at all. All of the subscriber's conferences are set to continue until the last participant disconnects.
Off	**Yes**	**Yes**	Continuation off. Subscriber can turn it on for the conference currently under way or for all future conferences.
		No	Continuation off. Subscriber can turn it on for all future conferences, but not for the conference currently under way.
	No	**Yes**	Continuation off. Subscriber can turn it on for the conference currently under way, but not for all future conferences.
		No	Continuation off. Subscriber can't turn it on at all. All of the subscriber's conferences are set to end when the subscriber disconnects.

If they may do so, subscribers can change the continuation behavior of all future conferences in two ways:

- By accessing the account options menu when dialing into the system (available if **Prompt for Menu** is turned on).
- By logging into the Moderator and accessing its account options settings.

If they may do so, subscribers can change the continuation behavior of the conference currently under way by entering the correct touchtone command (the default is ***8**).

Deleting a Subscriber Group

You can't delete a group that has active subscribers. First, you must change the subscribers to a different group. For instructions, see the *ReadiVoice Provisioning Guide*. You can't delete the last remaining subscriber group.

Caution! Be sure you want to permanently remove all information about a group before you delete it.

To delete a subscriber group:

1. In the navigation bar, click **Subscriber Groups**.

   The **Subscriber Group** page appears (Figure 2-16 on page 47).

2. Find the group you want to delete and click its **Delete** button (red "–").

   A dialog box asks you to confirm deleting the group.

3. Click **OK**.

   The system confirms that the group has been deleted.

Repeat the above procedure to delete another subscriber group.

Setting Up the Provisioning Interface

After working with the subscriber group settings, you may realize that your company doesn't need or want to use all the available fields when provisioning subscribers or modifying their accounts.

You can customize the Provisioning interface so that provisioners see only the fields you want them to use. You can also change the field names to match your company's usage.

The **Quick Provisioning Setup** page (Figure 2-21 and Figure 2-22) lets you decide what appears on the **Add Subscriber** and **Edit Subscriber** pages. You can:

- Hide fields and settings that your company doesn't need.

  A few fields are required by the system. They don't have check boxes by them, so you can't turn them off (but you can rename them). You probably want to require at least some subscriber and billing information.

  Other fields may be necessary because of your system configuration (such as **Subscriber External ID**) or to implement certain features (such as **Dial Out Prefix** and **Dial Out Postfix**).

- Change the names of fields and settings.

  When renaming fields and settings, keep in mind that space is limited. Check your changes to make sure they display properly.

  If you rename fields and settings that apply to subscriber groups, those changes also appear on the **Edit Group** page.

To customize what appears in the Provisioning interface:

1. In the **System Administration** navigation bar, click **Quick Provisioning Setup**.

   The **Quick Provisioning Setup** page appears.

2. To disable a field or setting (removing it from the Provisioning interface), click its check box to clear it. To re-enable a field or setting, click its check box again to select it.

Caution! Some ReadiVoice system configurations may require that certain fields and settings be available in Provisioning even though the Quick Provisioning Setup page allows you to leave them out. Be sure you know which fields are needed for your system configuration.

3. To change how a field or setting is labeled in the Provisioning interface, edit the name shown in its text box.

4 When you're finished making changes, click the **Update Quick Provisioning Settings** button.

The system confirms the provisioning setup changes and provides a link back to the **System Administration** page.

5 Test the changes you've made:

– Go to the Provisioning interface and add a test subscriber. Verify that the **Add Subscriber** page looks the way it should.

– Start a conference as the test subscriber and verify that all the features you want to support are available and operational.

Repeat this procedure if you need to make additional changes.

Figure 2-21 ***Quick Provisioning Setup** page (top)*

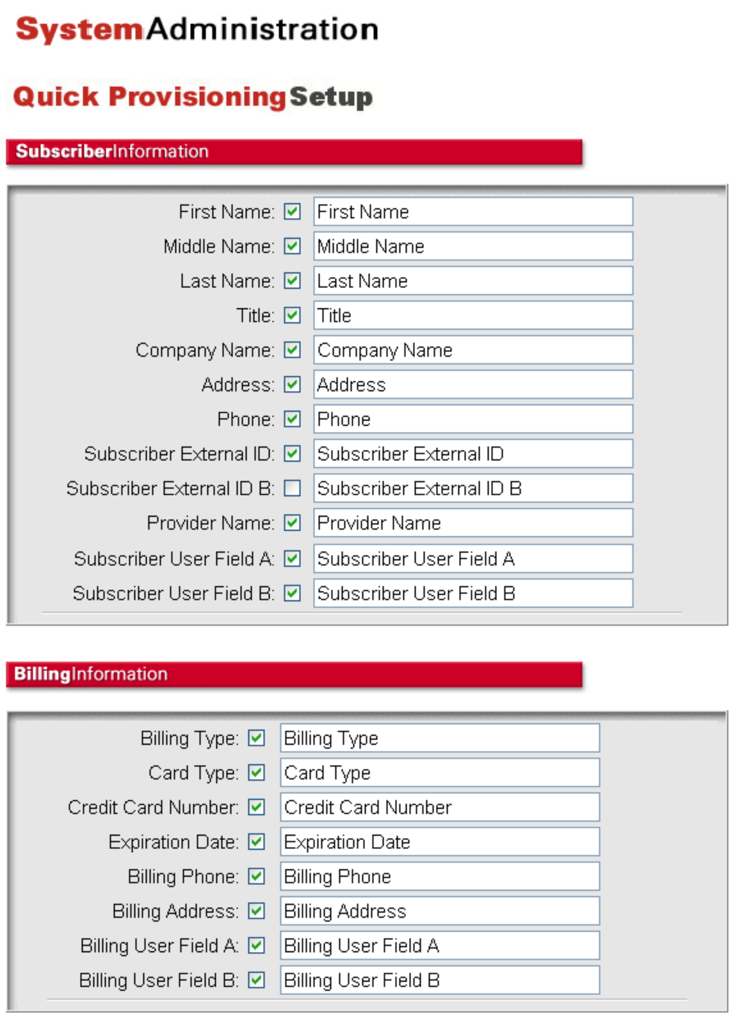
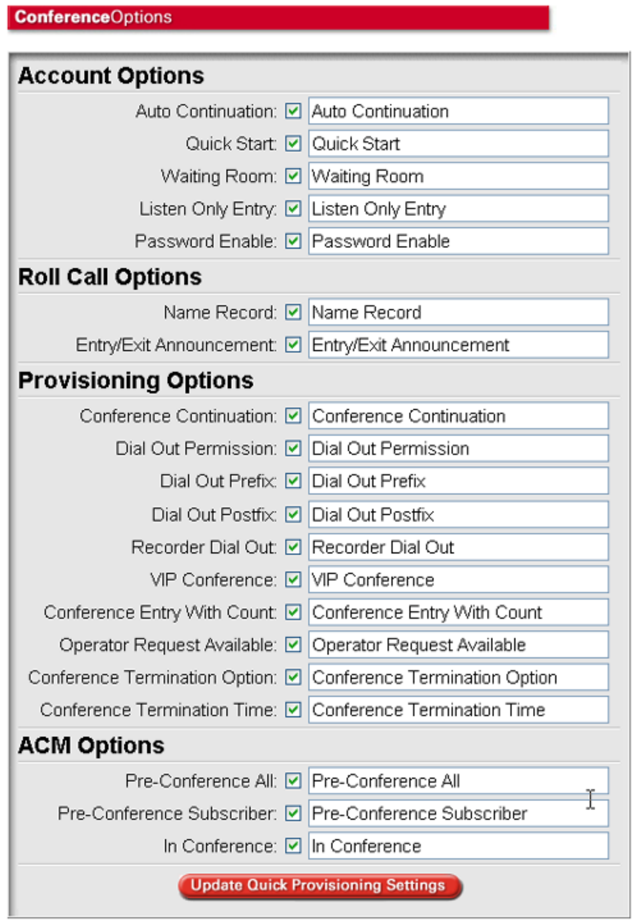


***Figure 2-22 Quick Provisioning Setup** page (bottom)*



Managing System Access

The ReadiVoice system supports several kinds of users with different roles and responsibilities. To give internal users (not subscribers) access to the system, you create entries for them on the **Passwords** page, assigning each a user name and password.

The user type (operator, provisioner, or administrator) controls which functions a user may access. A group provisioner is assigned to a single subscriber group and can add, change, or delete only subscribers within that group. A group operator is assigned to one or more subscriber groups and can monitor conferences and answer requests only from conferences within those subscriber groups.

Similarly, a group administrator can edit only a specific subscriber group and view only the CDRs from that group.

You can't add or delete group administrators directly. Instead, you do so when creating or editing the subscriber group. See "Working with Subscriber Groups" on page 46. You can, however, change a group administrator's password using the procedure described here.

The sections that follow describe how to add and delete internal users and change their passwords.

Adding an Internal User

1 In the **System Administration** navigation bar, click **Passwords**.

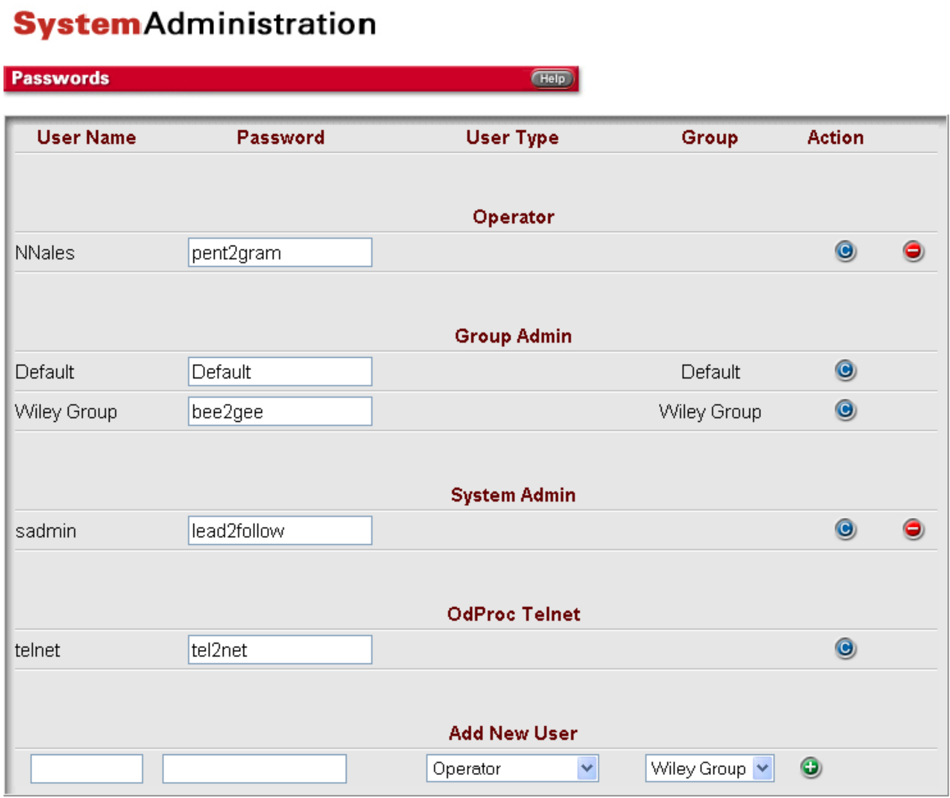
   The **Passwords** page appears (Figure 2-23). It lists the authorized users of the ReadiVoice system, showing the password, user type, and, if applicable, group for each. A blank row at the end of the list allows you to add a new user.

Caution! This page shows actual passwords, not asterisks, for all ReadiVoice users. Don't display it unless the monitor is secure from unauthorized personnel. Don't walk away from the monitor with this page displayed.

2 In the **User Name** field of the blank row, enter the login name for the new user. The maximum length is 30 characters.

3 In the **Password** field, enter the new user's password. The maximum length is 20 characters.

Figure 2-23 ***Passwords*** *page*



4 In the **User Type** list, select the user type.

The user types are operator, provisioner, group operator, group provisioner, and system administrator.

5 If the new user is a group provisioner or group operator, select from the **Group** list the subscriber group to which this user has access.

For a group operator, you can add or change subscriber groups later (see step 8). For a group provisioner, the group selected here is the only one to which the provisioner has access and it can't be changed.

6 Click the **Add** button.

The system confirms that the user has been added and provides a link for returning to the **Passwords** page.

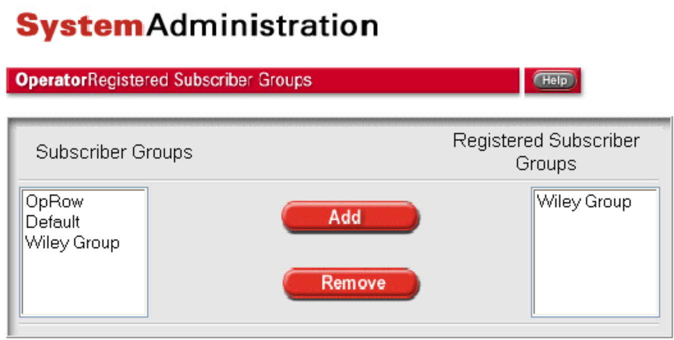
7 Click the link to reload the **Passwords** page.

The new entry appears at the end of the list, above the blank row.

8 If the new user is a group operator and you want to add more subscriber groups, click the new entry's **Register Groups** link.

The **Operator Registered Subscriber Groups** page appears (Figure 2-24).

***Figure 2-24 Operator Registered Subscriber Groups** page*



9 Use the **Add** and **Remove** buttons to move selected groups into or out of this operator's **Registered Subscriber Groups** list.

10 When you're finished setting up this group operator, click **Passwords** in the navigation bar to return to the **Passwords** page.

Repeat the above procedure to add another user.

Changing an Internal User's Password

1 In the navigation bar, click **Passwords**.

The **Passwords** page appears (Figure 2-23). It lists the authorized users of the system, showing the password, user type, and, if applicable, group for each. Only the password can be modified.

To change a user's login name, user type, or group, delete the existing entry and add a new one. The **Passwords** page also lists the Telnet login for the system. You can change the password for this login, but can't delete it.

2 Locate the user whose password you want to change.

3 Replace or edit the existing password and click the **Commit** button (blue "c").

The system confirms that the password has been modified and provides a link for returning to the **Passwords** page.

4 Click the link to reload the **Passwords** page.

The user you modified has a new password.

Repeat the above procedure to change another password.

Deleting an Internal User

1. In the navigation bar, click **Passwords**.

   The Passwords page appears (Figure 2-23).

2. Locate the entry for the user you want to remove and click its **Delete** button (red "–").

   The system confirms that the user has been deleted and provides a link for returning to the **Passwords** page.

3. Click the link to reload the **Passwords** page.

   The entry you deleted is gone from the list.

Repeat the above procedure to delete another user.

Defining Invalid Subscriber Passwords

As a security feature, the ReadiVoice system lets you specify certain number combinations as not valid subscriber passwords. For instance, you can disallow combinations such as 1111 and 1234.

The ReadiVoice system stores the invalid combinations in the invalid subscriber password table. You can add entries to and delete entries from this table. Adding entries has no effect on subscribers currently using the entries. It only prevents these entries from being chosen in the future.

Adding Entries to the List of Invalid Subscriber Passwords

1. In the **System Administration** navigation bar, click **Invalid Passwords**.

   The **Invalid Subscriber Password Table** page appears (Figure 2-25).

2. In the **Enter a Password to Add to the List** field, type the number you want to invalidate as a subscriber password.

3. Click the **Add** button (green "+").

   The system confirms that the entry has been added and provides a link for returning to the **Invalid Subscriber Password Table** page.

4. Click the link to reload the **Invalid Subscriber Password Table** page.

   The entry you added appears in the list.

Repeat the above procedure to add another invalid password entry.

Deleting Entries from the List of Invalid Subscriber Passwords

1 In the navigation bar, click **Invalid Subscriber Passwords**.

The **Invalid Subscriber Password Table** page appears (Figure 2-25).

2 Scroll through the list of invalid subscriber passwords and select the number you want to delete.

3 Click the **Delete** button (red "–").

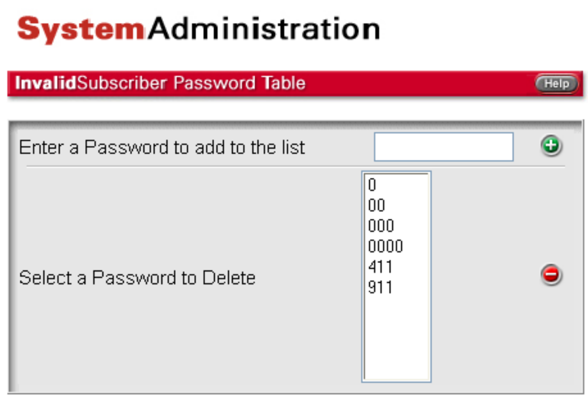
The system confirms that the entry has been deleted and provides a link for returning to the **Invalid Subscriber Password Table** page.

4 Click the link to reload the **Invalid Subscriber Password Table** page.

The entry you deleted is gone from the list.

Repeat the above procedure to delete another entry.

***Figure 2-25 Invalid Subscriber Password Table** page*



3

Monitoring the ReadiVoice System

This chapter describes the ReadiVoice system monitoring functions that you can access with your browser. You can view the overall status of your system, usage levels, bridge and span status, operator data, and conference data.

Accessing the Monitoring Tools

You can access the ReadiVoice system monitoring tools from any computer that can connect to the ReadiVoice system's Web server and has a compatible Web browser (Microsoft Internet Explorer, version 5.5 or later).

Some functions also require a Java plug-in (Java Runtime Engine, or JRE). If you don't have the necessary plug-in, your browser should ask you if you want to download and install it. You must do so to use those functions. Polycom has tested these Java applets with the Sun Microsystems Java Runtime Engine (JRE) 1.3 on Microsoft Windows 2000.

These applets should work on other platforms and operating systems and with newer versions of the JRE, but that depends on proper implementation by the operating system and Java virtual machine. Polycom hasn't tested other possible combinations and can't be responsible for implementation or compatibility issues beyond our control.

Access to the monitoring tools is restricted to authorized users. You must know the correct user name and password.

To access ReadiVoice system monitoring tools:

1 Point your browser to the ReadiVoice home page for your system (for example, http://rvoice.com/index2.html).

   The ReadiVoice internal home page appears (Figure 3-1). The navigation bar is on the right.

2 Select the link for the information you want to see:

   – Click **Show System Status** to see information about disk usage and running processes on your CACS server.

– Click **Show Conference Status** to see information about current conferences.

– Click **Show Conference Report** to view call detail record (CDR) data for completed conferences sorted by subscriber.

– Click **Show CDR Data** to view CDR data sorted by conference.

– Click **Operator Statistics** to view data taken from requests for operator assistance, or to view statistics calculated from the data.

– Click **SNMP Monitor** to view system statistics in real time.

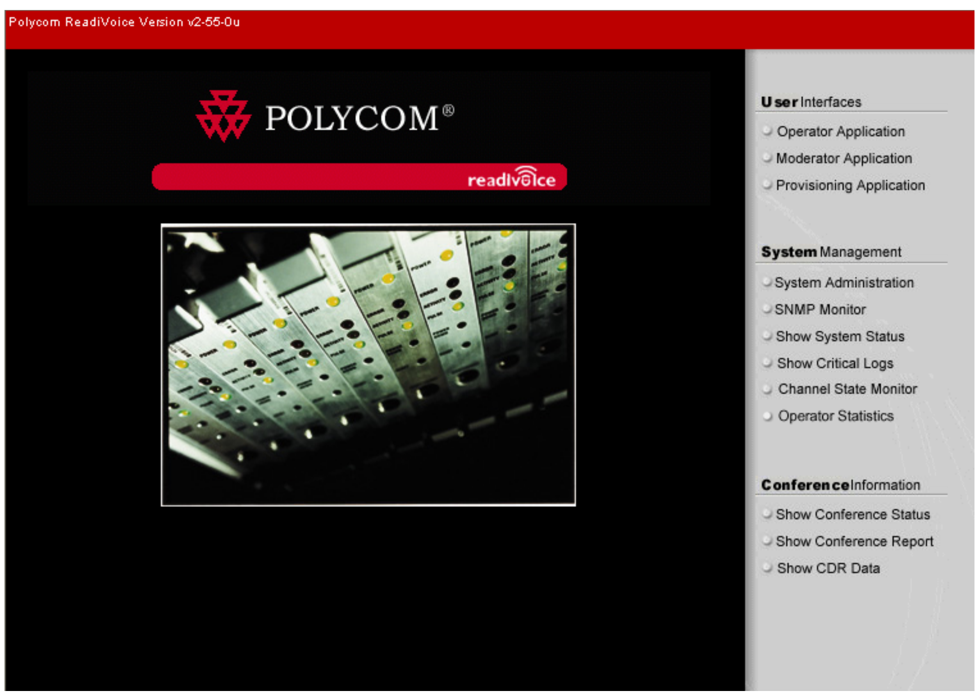
– Click **Channel State Monitor** to see real-time display of the lines in use on each bridge (not available in an IP system).

– Click **Show Critical Logs** to check for critical events in the bridge log files (described in Chapter 6, "Diagnostics and Troubleshooting," on page 145).

If you aren't already logged in as a system administrator, the login dialog box appears.

3 Enter a user name and password authorized for administrative access. Then click **OK**.

Figure 3-1 ReadiVoice internal user home page



Checking System Status

The **System Status** page (Figure 3-2) appears when you click the **Show System Status** link in the navigation bar.

The **System Disk Usage** and **Informix Disk Usage** panels show disk usage statistics for your CACS server's Solaris partitions and Informix chunks. For each partition or chunk, they show:

KB Size — The total size of the partition or chunk in kilobytes.

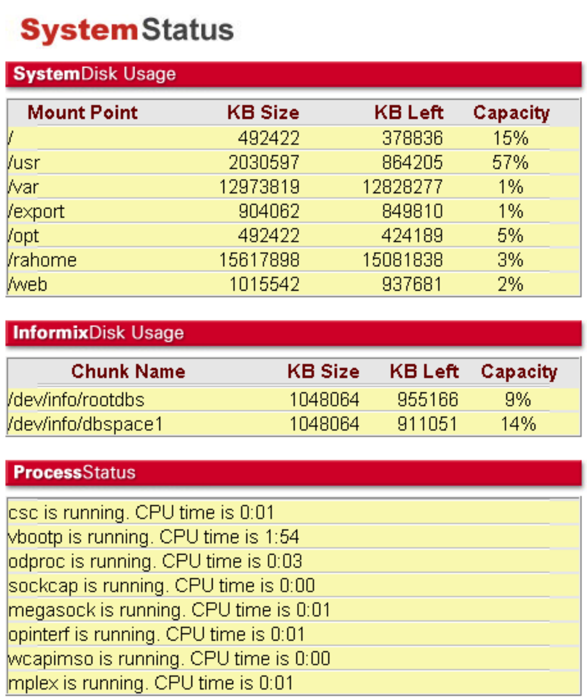
KB Left — The amount of free space in kilobytes.

Capacity — The percentage of the total size that is used.

The **Process Status** panel shows the status (running or not running) and CPU time of the ReadiVoice server processes. The processes listed depend on whether this is a PSTN or IP system and how it's configured. See "Stopping and Restarting ReadiVoice Processes" on page 111.

Caution! ReadiVoice behavior is unpredictable unless all server processes are running.

Figure 3-2 ***System Status** page*



SystemStatus

SystemDisk Usage

Mount Point	KB Size	KB Left	Capacity
/	492422	378836	15%
/usr	2030597	864205	57%
/var	12973819	12828277	1%
/export	904062	849810	1%
/opt	492422	424189	5%
/rahome	15617898	15081838	3%
/web	1015542	937681	2%

InformixDisk Usage

Chunk Name	KB Size	KB Left	Capacity
/dev/info/rootdbs	1048064	955166	9%
/dev/info/dbspace1	1048064	911051	14%

ProcessStatus

csc is running. CPU time is 0:01
vbootp is running. CPU time is 1:54
odproc is running. CPU time is 0:03
sockcap is running. CPU time is 0:00
megasock is running. CPU time is 0:01
opinterf is running. CPU time is 0:01
wcapimso is running. CPU time is 0:00
mplex is running. CPU time is 0:01

Viewing Conferencing Information

The **Conference Information** links in the navigation bar let you look at:

- Information about the conferences currently running.
- Summary CDR data, with one entry per subscriber, for a time period you choose (up to one month).
- Summary CDR data, with one entry per conference.
- Detailed CDR data for a specific conference.

Call detail records (CDRs) are used for billing and provide detailed information about a conference, its participants, and all features used and events associated with it. If you're a system administrator, you can look at any CDR data in the database; if you're a group administrator or group provisioner, you can look only at the CDR data for your subscriber group.

For descriptions of the CDR data, see Appendix B.

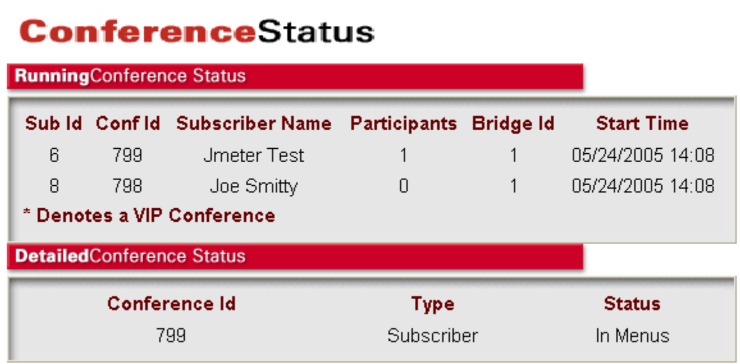
Current Conferences Information

The **Conference Status** page appears when you click the **Show Conference Status** link in the navigation bar. It provides an overview of the current conferencing activity on the system.

The **Conference Status** page shows:

- At the top, summary information about all currently running conferences.
- Below the summary table, status information for each running conference, showing details about each line in the conference.

Figure 3-3 ***Conference Status*** *page*



CDR Data by Subscriber

When you click the **Show Conference Report** link in the navigation bar, the **CDR Data** page displays CDR data by subscriber. Each entry in the **Conference Report** panel shows the aggregate conferencing data for one subscriber.

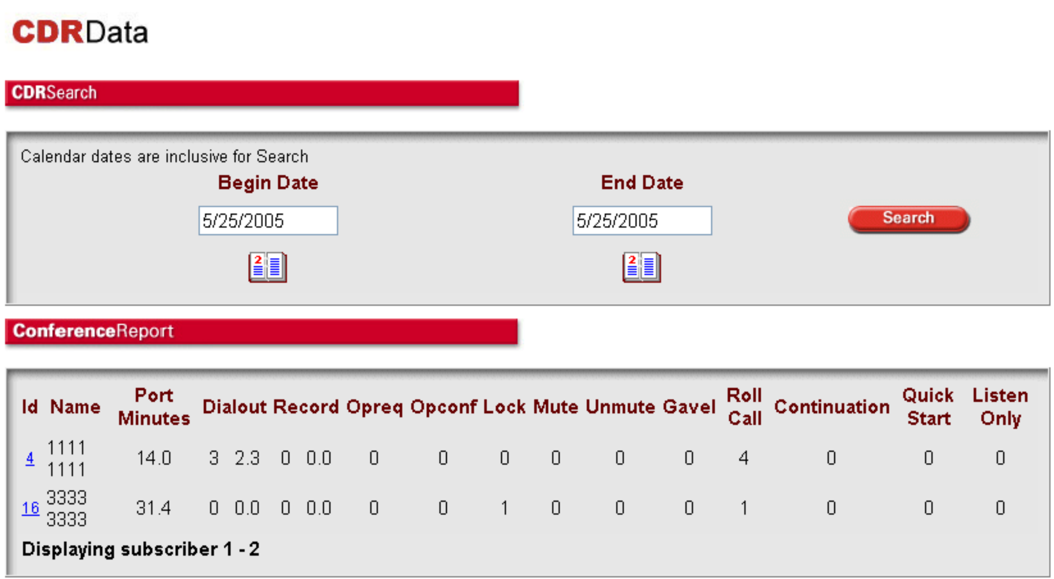
When the page first appears, it shows the subscriber data for the current date. To see subscriber data for a different date or range of dates, change **Begin Date**, **End Date**, or both, and click **Search**.

For each subscriber listed, the report shows:

- Subscriber ID and name.
- Total number of port minutes.
- Number of dial-outs and dial-out minutes.
- Number of times each billable feature was used.

To see more detail for a subscriber, click that subscriber's ID in the list. The system displays summary CDR data for the subscriber's conferences, with one entry per conference. This is the same display that you get by clicking **Show CDR Data** and then searching for that subscriber ID and date range. See the next section.

Figure 3-4 *Viewing CDR data by subscriber*



CDR Data by Conference

When you click the **Show CDR Data** link in the navigation bar, the **CDR Data** page presents an expanded search panel (Figure 3-5), where you can specify various criteria, such as subscriber ID or name, and set a date range.

When you click **Search**, the system retrieves data for the conferences within your date range that match your search criteria. Each entry in the **Conference Report** panel shows the data for one conference. Figure 3-5 shows the results of a search for a specific subscriber ID.

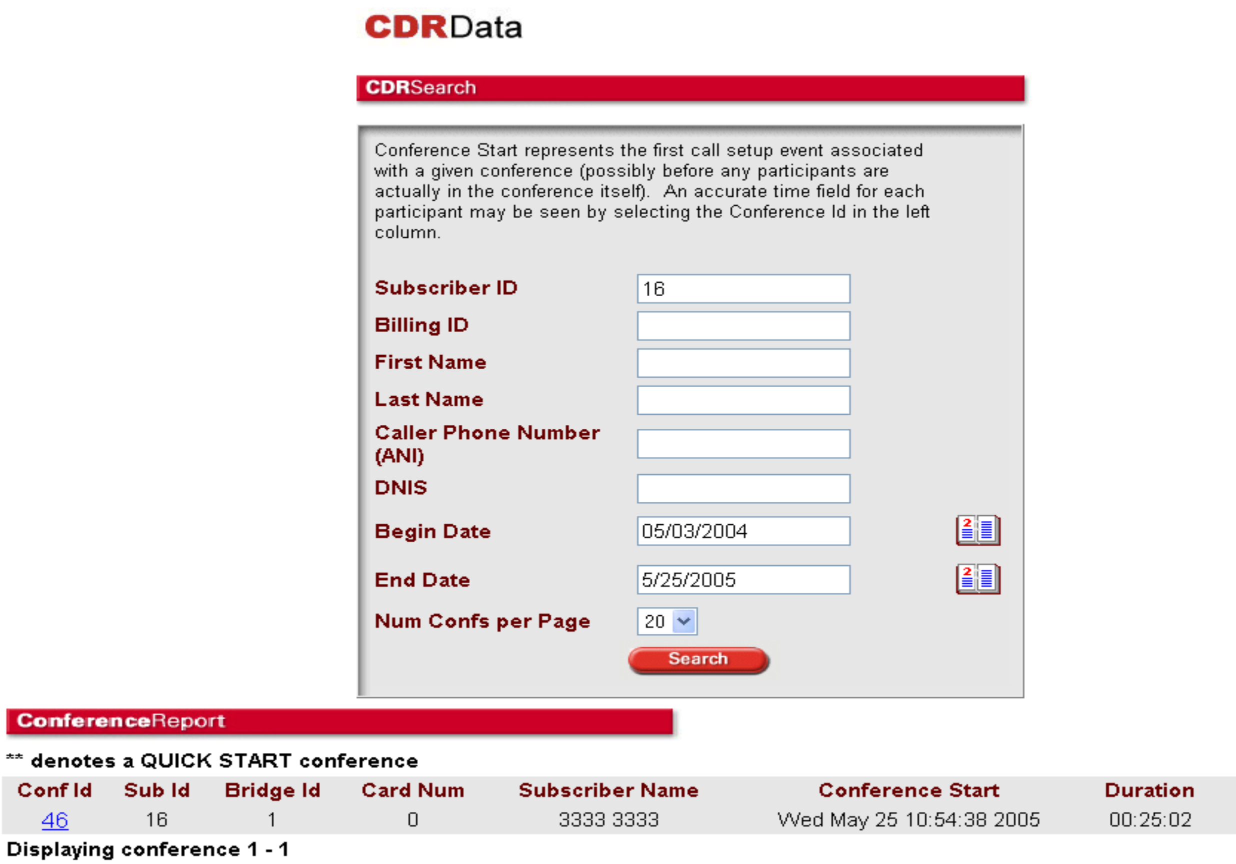
If the date range includes today, the list may include conferences that are currently running or whose CDRs haven't been fully processed yet; these records will be incomplete.

For each listed conference, the **Conference Report** panel shows:

- Conference and subscriber IDs.
- Bridge ID and card number on which conference ran.
- Subscriber's name.
- Start time and duration (or **Incomplete** if the conference is still running).
- Number of participants (or **N/A** if the conference is still running).

To see details for a specific conference, click its conference ID. The **CDR Data** page displays detailed CDR data for the conference. See the next section.

Figure 3-5 Retrieving CDR data by conference



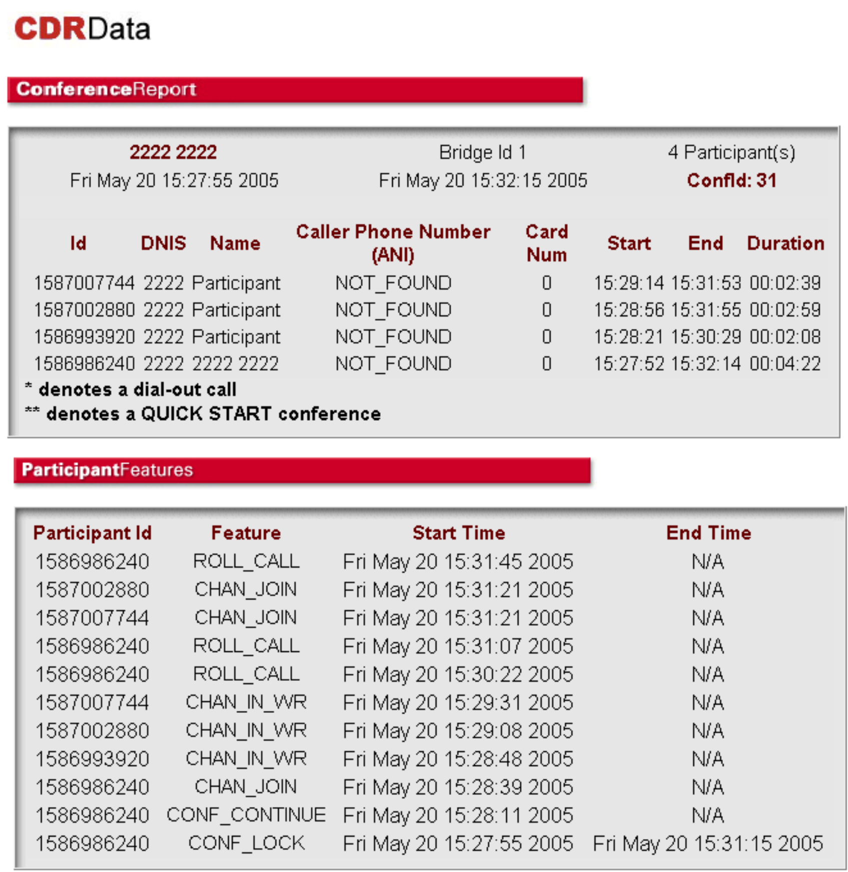
Detailed CDR Data for One Conference

While viewing CDR data by conference, click a conference ID for more information about that conference. The **CDR Data** page displays detailed CDR data for the conference (Figure 3-6). The **Conference Report** panel at the top contains information about the conference and its participants. The **Participant Features** panel contains records of billable feature usage. For descriptions of the features that may be listed, see Table B-3, "Feature Information (cdr_post_state)" on page 258.

If the conference is still running or its CDRs haven't been fully processed yet, these records will be incomplete.

If you have a non-routed system using private (not shared) access numbers, and you're using hidden numbers, then the **Conference Report** panel shows the hidden numbers instead of the DNIS numbers.

Figure 3-6 *Viewing detailed CDR data for a conference*



CDRData

ConferenceReport

2222 2222	Bridge Id 1	4 Participant(s)
Fri May 20 15:27:55 2005	Fri May 20 15:32:15 2005	ConfId: 31

Id	DNIS	Name	Caller Phone Number (ANI)	Card Num	Start	End	Duration
1587007744	2222	Participant	NOT_FOUND	0	15:29:14	15:31:53	00:02:39
1587002880	2222	Participant	NOT_FOUND	0	15:28:56	15:31:55	00:02:59
1586993920	2222	Participant	NOT_FOUND	0	15:28:21	15:30:29	00:02:08
1586986240	2222	2222 2222	NOT_FOUND	0	15:27:52	15:32:14	00:04:22

* denotes a dial-out call
** denotes a QUICK START conference

ParticipantFeatures

Participant Id	Feature	Start Time	End Time
1586986240	ROLL_CALL	Fri May 20 15:31:45 2005	N/A
1587002880	CHAN_JOIN	Fri May 20 15:31:21 2005	N/A
1587007744	CHAN_JOIN	Fri May 20 15:31:21 2005	N/A
1586986240	ROLL_CALL	Fri May 20 15:31:07 2005	N/A
1586986240	ROLL_CALL	Fri May 20 15:30:22 2005	N/A
1587007744	CHAN_IN_WR	Fri May 20 15:29:31 2005	N/A
1587002880	CHAN_IN_WR	Fri May 20 15:29:08 2005	N/A
1586993920	CHAN_IN_WR	Fri May 20 15:28:48 2005	N/A
1586986240	CHAN_JOIN	Fri May 20 15:28:39 2005	N/A
1586986240	CONF_CONTINUE	Fri May 20 15:28:11 2005	N/A
1586986240	CONF_LOCK	Fri May 20 15:27:55 2005	Fri May 20 15:31:15 2005

Viewing Operator Information

The **Operator Statistics** page appears when you click the **Operator Statistics** link in the navigation bar of the ReadiVoice internal user home page. From here, you can:

- Click **View Requests** in the navigation bar to see records of operator requests (for all operators, a specific operator, or a specific subscriber). Follow the procedure in "Viewing Operator Request Records" on page 77.
- Click **View Statistics** in the navigation bar to see operator request statistics (for all operators, a specific operator, or a specific subscriber). Follow the procedure in "Viewing Operator Request Statistics" on page 79.

Operator request data is available only if operator data logging is enabled on the **System Configuration** page (see "Changing System Configuration Settings" on page 118). If it is, the ReadiVoice system creates an operator log file each night at midnight containing all the operator request records for the preceding day.

The **Operator Statistics** pages let you retrieve and view data from these log files The operator logs are deleted after 31 days, so you can see operator request data for up to a month.

Viewing Operator Request Records

1. In the **Operator Statistics** page navigation bar, click **View Requests**.

   The **Requests** page appears (Figure 3-7). It provides you with three scope options in the navigation bar (**For All Operators**, **For Specific Operator**, and **For Specific Subscriber**).

2. In the navigation bar, click the link that corresponds with the scope of the records you want to retrieve.

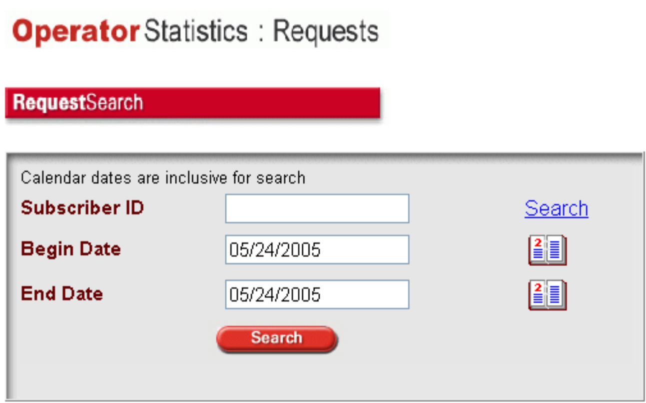
   The **Requests Search** page displays search fields that vary slightly depending on the choice you made. Figure 3-8 shows the search fields that appear when you choose **For Specific Subscriber**.

3. If you chose **For Specific Operator** or **For Specific Subscriber**, select the operator from the list or enter the subscriber's ID (if you don't know the subscriber's ID, click the **Search** link to its right to search for the subscriber).

4. If you chose **For All Operators**, select the number of records you want to see per page.

Figure 3-7 *The **Operator Statistics Requests** page*

Figure 3-8 *Searching for operator requests by subscriber*



5 Set the **Begin Date** and **End Date** fields by doing one of the following:

– Click in the field and edit the date.

– Click the field's calendar icon and click a date in the calendar that appears.

Regardless of which dates you choose, records are only available for the past 31 days and aren't available for the current day (the system collects the day's data at midnight).

6 Click the **Search** button.

The ReadiVoice system displays the answered requests matching your search criteria. For each request, it shows:

– Type of operator request (private or conference).

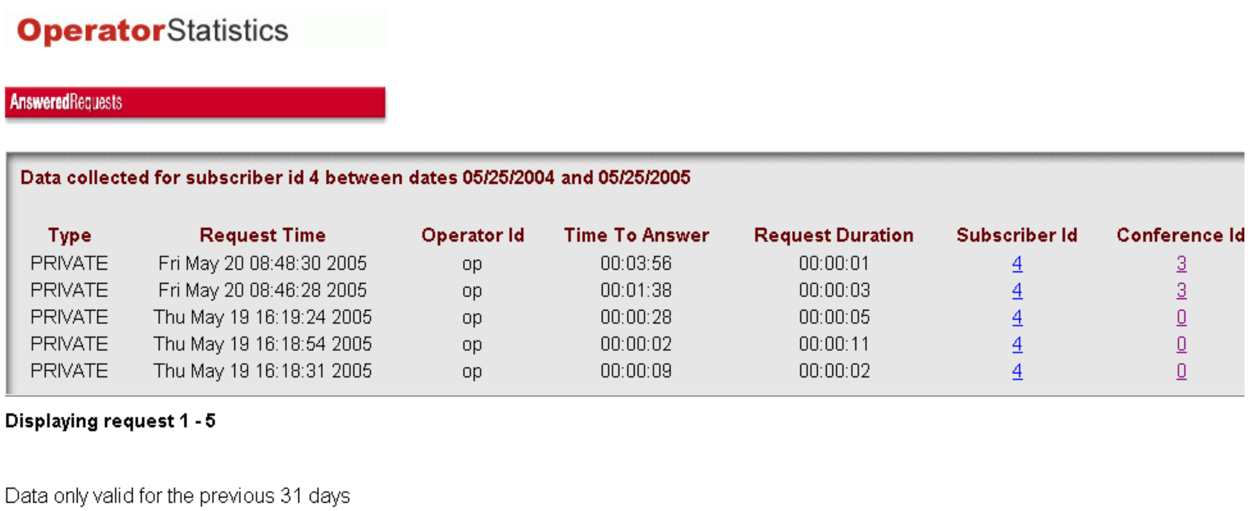
– Date and time the request was made.

– Operator ID of the operator who answered the request.

– How long it took an operator to answer the request.

- How long the operator spent on the request (talking with the subscriber or conference).
- Subscriber ID for the conference in question. Click the ID to retrieve a conference report for that subscriber.
- Conference ID of the conference. Click the ID to retrieve a detailed report for that conference.

Figure 3-9 *Results of operator request search*



OperatorStatistics

AnsweredRequests

Data collected for subscriber id 4 between dates 05/25/2004 and 05/25/2005

Type	Request Time	Operator Id	Time To Answer	Request Duration	Subscriber Id	Conference Id
PRIVATE	Fri May 20 08:48:30 2005	op	00:03:56	00:00:01	4	3
PRIVATE	Fri May 20 08:46:28 2005	op	00:01:38	00:00:03	4	3
PRIVATE	Thu May 19 16:19:24 2005	op	00:00:28	00:00:05	4	0
PRIVATE	Thu May 19 16:18:54 2005	op	00:00:02	00:00:11	4	0
PRIVATE	Thu May 19 16:18:31 2005	op	00:00:09	00:00:02	4	0

Displaying request 1 - 5

Data only valid for the previous 31 days

Viewing Operator Request Statistics

1. In the **Operator Statistics** page navigation bar, click **View Statistics**.

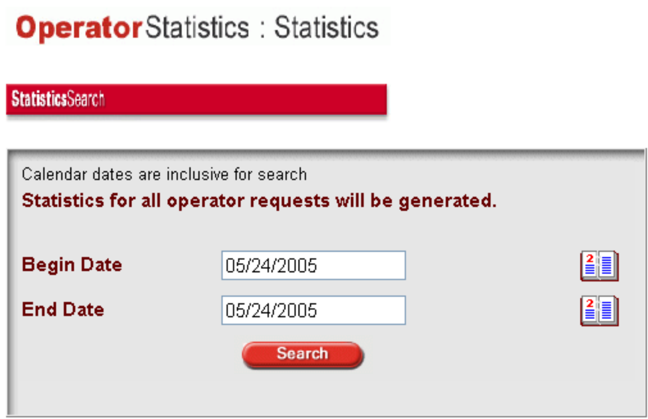
   The **Statistics** page appears. It looks similar to the **Requests** page (Figure 3-7) and has the same three options in the navigation bar (**For All Operators**, **For Specific Operator**, and **For Specific Subscriber**).

2. In the navigation bar, click the link that corresponds with the statistics you want to generate.

   The **Statistics Search** page displays search fields that vary slightly depending on the choice you made. Figure 3-10 shows the search fields that appear when you choose **For All Operators**.

3. If you chose **For Specific Operator** or **For Specific Subscriber**, select the operator from the list or enter the subscriber's ID (if you don't know the subscriber's ID, click the **Search** link to its right to search for the subscriber).

Figure 3-10 Generating statistics for all operators



4 Set the **Begin Date** and **End Date** fields by doing one of the following:

- Click in the field and edit the date.
- Click the field's calendar icon and click a date in the calendar that appears.

Regardless of which dates you choose, records are only available for the past 31 days and aren't available for the current day (the system collects the day's data at midnight).

5 Click the **Search** button.

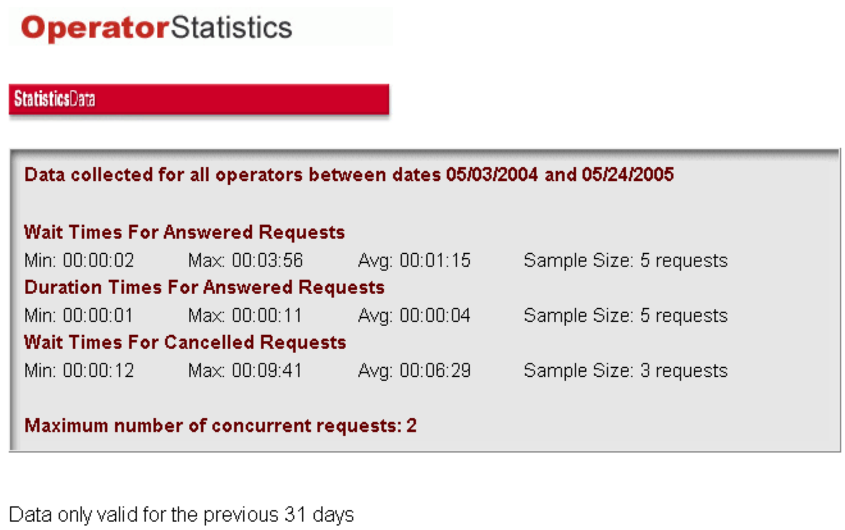
The ReadiVoice system displays operator request statistics for the operator, subscriber, and/or time period matching your search criteria (Figure 3-11). It shows:

- Minimum, maximum, and average wait for request to be answered.
- Minimum, maximum, and average time spent on a request.
- Sample sizes (number of requests included in statistics).

If you requested statistics for all operators, it also shows:

- Minimum, maximum, and average wait for canceled requests.
- Maximum number of requests in the queue at any one time.

Figure 3-11 Results of generating statistics for all operators



Using the SNMP Monitor

The ReadiVoice system supports the Simple Network Management Protocol (SNMP). SNMP lets you monitor the ReadiVoice system with standard network administration tools, such as HP Openview from Hewlett Packard or similar tools from companies such as SUN or IBM.

documents the system and bridge SNMP events. "Enabling SNMP Logging" on page 141 describes SNMP logging, which writes SNMP data to log files on an ongoing basis.

But, if you just want to check system statistics in real time, you can use SNMP Monitor. SNMP Monitor is a Java applet that you run from your browser. It provides a quick and convenient way to monitor the contents of the SNMP Management Information Base (MIB).

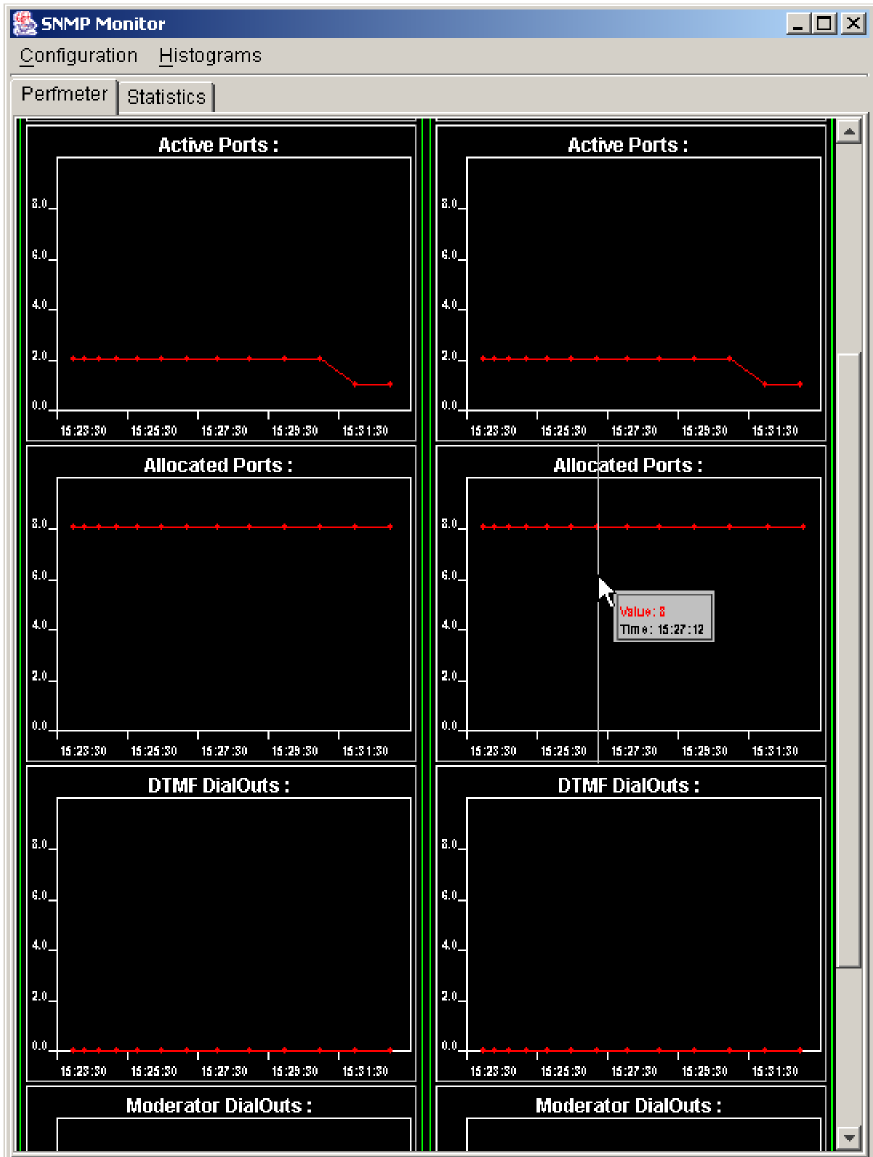
To run SNMP Monitor:

1 On the ReadiVoice home page, click the **SNMP Monitor** link.

The Java applet loads, which may take a few moments. Then, the **SNMP Monitor** window appears. The **Perfmeter** tab is selected, and it starts graphing selected MIB data over time, both system-level and for each individual bridge. See Figure 3-12.

Although the SNMP Monitor applet opens in its own window, your browser must remain active and pointed to the Monitor host page. If you close your browser or navigate away from the host page, SNMP Monitor closes.

***Figure 3-12 Perfmeter** tab of **SNMP Monitor** window*



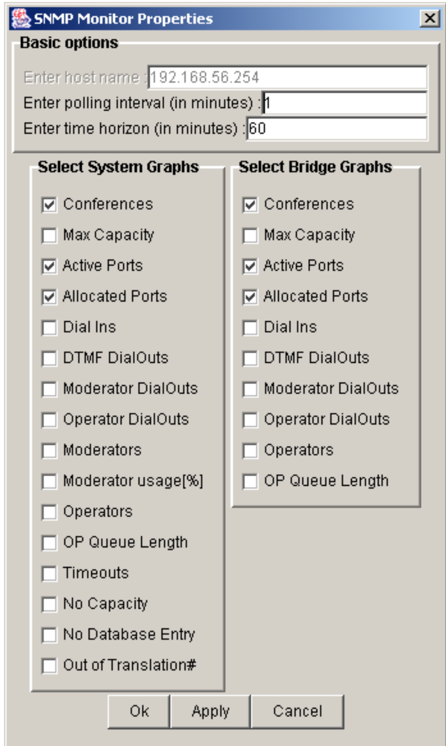
2 Point your mouse pointer at a specific data point on a graph.

A small window displays the MIB data value and time for that point on the graph.

3 To customize which MIB elements are monitored, select **Setup** from the **Configuration** menu.

The **SNMP Monitor Properties** dialog box opens (Figure 3-13).

Figure 3-13 ***SNMP Monitor Properties*** *dialog box*



4. Change the polling interval and time horizon, if you want, and choose the graphs you want displayed. Then click **Apply**.

   The **SNMP Monitor** window reflects the changes you made.

5. Resize the **SNMP Monitor** window, if necessary, to display the graphs you selected at a convenient size.

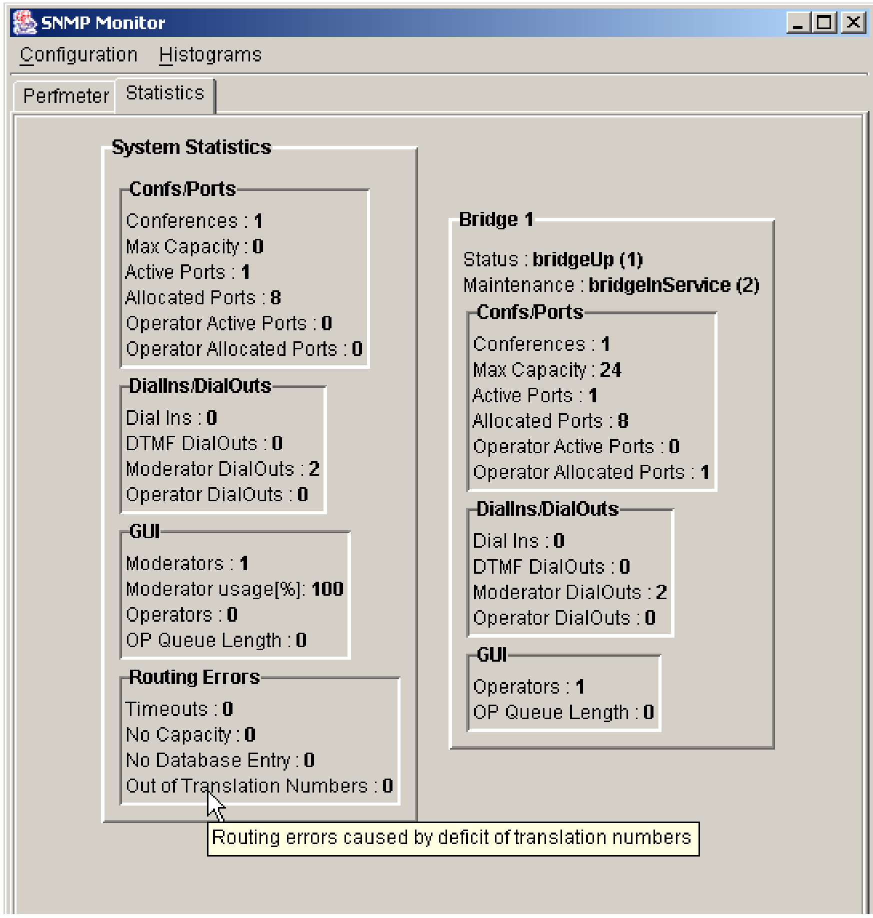
6. Make additional changes, if you want, in the **SNMP Monitor Properties** dialog box. When you're finished, click **OK**.

   The **SNMP Monitor Properties** dialog box closes.

7. To see all MIB statistics for your system, click the **Statistics** tab.

   The **Statistics** tab displays, in text form, all MIB elements, both for the system and for each individual bridge (Figure 3-14).

***Figure 3-14 Statistics** tab of **SNMP Monitor** window*



8 To see a record of the system's conference and port usage levels over a longer time period, select **Histograms** from the **Histograms** menu.

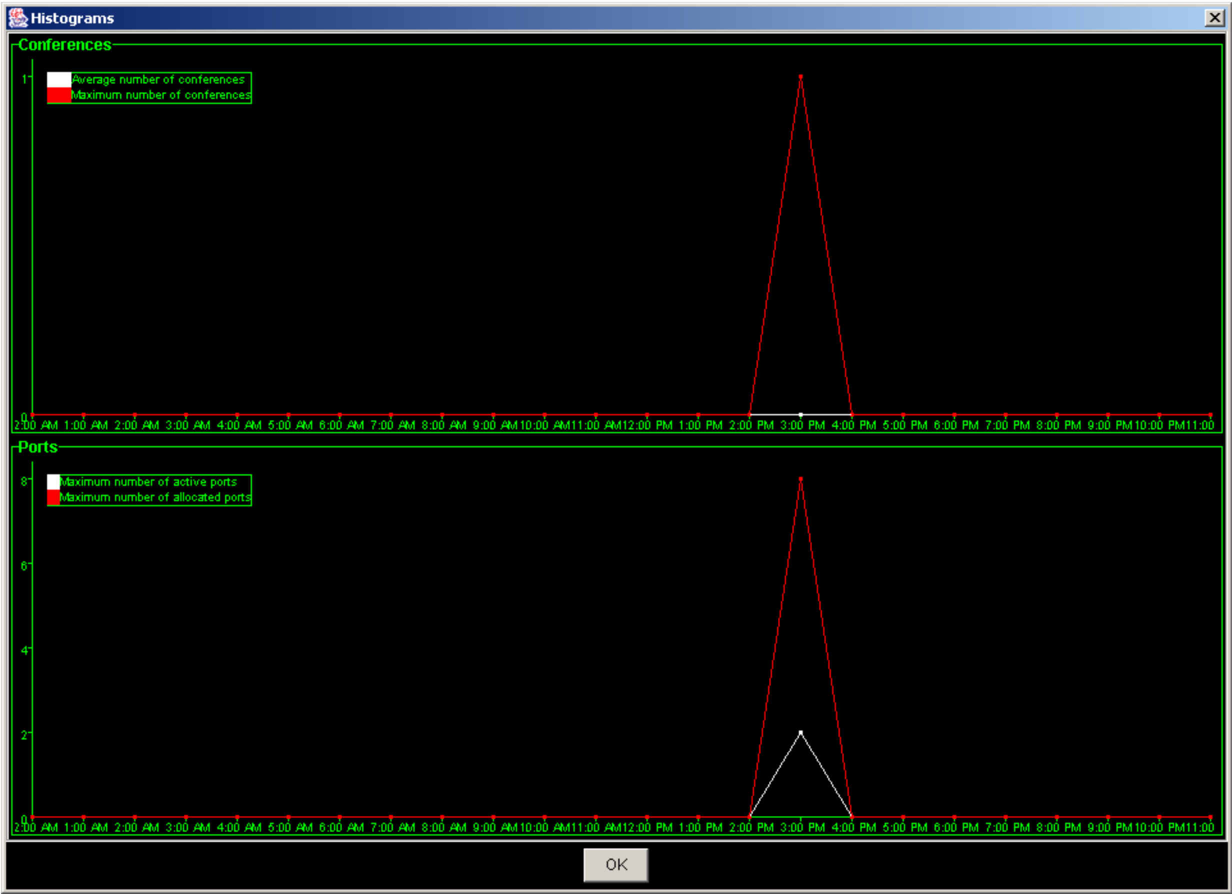
The **Histograms** window appears (Figure 3-15). It displays conference and port usage levels over time.

The time period shown on the horizontal axis is always 24 hours, from midnight to midnight. Therefore, the data to the left of the current hour is from today and the data to the right of the current hour is from yesterday. The MIB data is updated hourly.

9 To close the **Histograms** window, click **OK**.

10 To stop running SNMP Monitor entirely, select **Exit** from the **SNMP Monitor** window's **Configuration** menu, close your browser, or navigate away from the host page.

Figure 3-15 ***Histograms*** *window of SNMP Monitor*



Using the Channel State Monitor

In a PSTN system, you can use the Channel State Monitor tool to monitor the teleconferencing bridge resources of the system (cards, spans, and channels) and to enable or disable telephony spans.

To run Channel State Monitor:

1 In the navigation bar of the internal user home page, click **Channel State Monitor**. If prompted, enter your system administration user name and password.

   The initial **Channel State Monitor** page appears (Figure 3-16).

2 Click the **Display** button (purple "D") of the bridge for which you want to see span and channel information.

   The Java applet loads, which may take a few moments, and displays summary status information for all the processor (VCE or HMod) cards in the bridge (Figure 3-17). The cable icon in the lower-right corner shows the state of the network connection between the CACS and the bridge; it blinks if the connection is lost.

Figure 3-16 ***Channel State Monitor*** *page, listing bridges in the system*

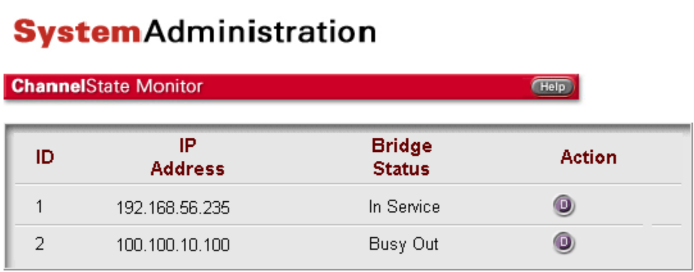


Figure 3-17 Channel State Monitor's summary information for a bridge

3 Click on a card to see detailed information about that card.

Channel State Monitor opens an applet window for the card (Figure 3-18). The top row contains a status indicator for each span:

Green — span is up and connected.

Red — span is in red alarm (connection to far end lost).

Yellow — span is in yellow alarm.

Blue — span is in blue alarm (busied out).

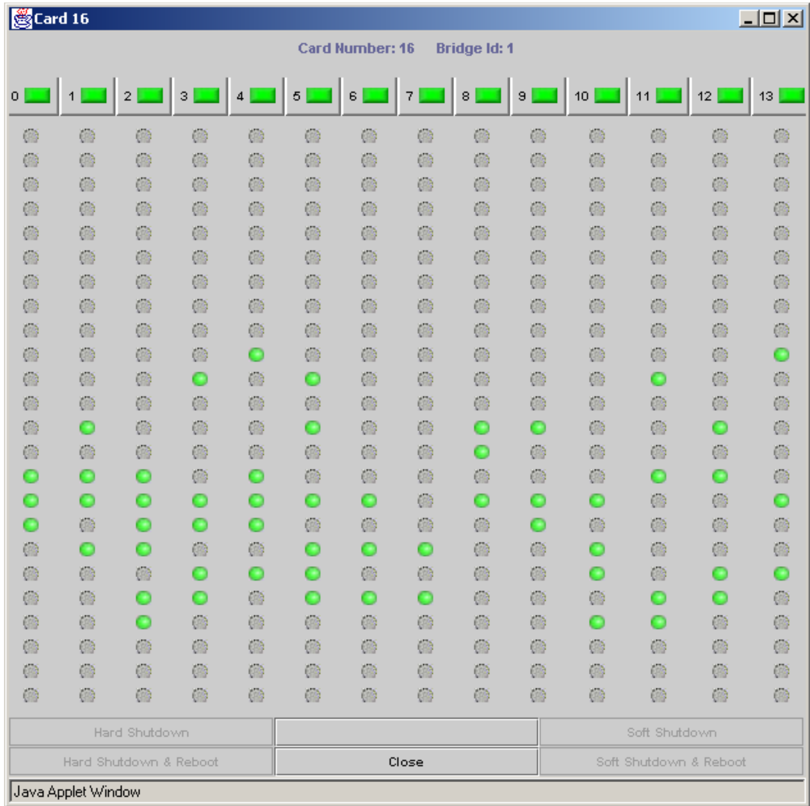
Black (or dark gray) — span has been disabled.

Below each span indicator is a column of status indicators for its channels:

Green — channel is off hook.

Gray — channel is on hook.

Figure 3-18 *Applet window with details for the selected card*



4 To see additional information about a span, click its status indicator in the card window.

Another applet window appears, displaying configuration and status details for the selected span (Figure 3-19). The buttons on the right let you change the span's status:

Hard Span Disable — Takes the span out of service.

Enable Span — Restores a disabled span to service. This takes some time (typically, about 30 seconds).

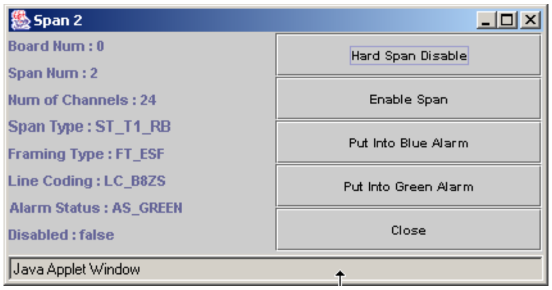
Put Into Blue Alarm — Busies out the span.

Put Into Green Alarm — Restores a busied-out span to service. This takes some time (typically, about 30 seconds).

Close — Closes the applet window for the span.

Caution! Disabling a span or putting it into blue alarm immediately hooks all channels on that span.

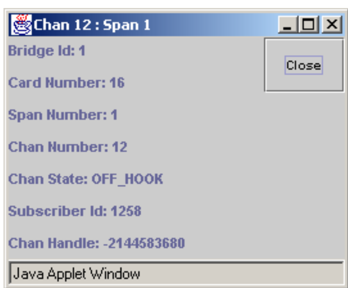
Figure 3-19 Applet window with details for the selected span



5 To see additional information about a channel, click its status indicator in the card window.

Another applet window appears, displaying configuration and status details for the selected channel (Figure 3-20). If the channel is off hook, the information includes the handle of the channel process and the subscriber ID of the associated conference process, if any.

Figure 3-20 Applet window with details for the selected channel (port)



4

Maintaining the ReadiVoice System

This chapter describes how to maintain your ReadiVoice Intelligent Voice Conferencing System. It includes information about backup procedures and other routine maintenance tasks. It assumes that you are knowledgeable about Solaris UNIX, Sun servers, Informix databases, and networking.

Caution! The maintenance procedures are essential to proper operation of the ReadiVoice system. Failure to perform these tasks as recommended may result in critical failure of the ReadiVoice system.

When the Polycom installs and configures your system, we set up certain automated processes, including transaction logging for the Informix database and monthly purging of ReadiVoice log files.

Nevertheless, you must perform some maintenance tasks on a regular basis in order to check your system, keep it operating properly, and ensure that you can recover from a hardware or software failure. These tasks are described in this chapter. Topics include:

- "Quick Maintenance Checklist" on page 92 summarizes the daily and weekly tasks described in more detail later.
- "Understanding Informix Backup" on page 93 provides an overview of archiving and logical logging, and helps you understand backup strategies and options.
- "Backing Up Logical Logs" on page 96 describes how to make sure that Informix Dynamic Server's transaction logs are being backed up to tape on an ongoing basis.
- "Daily Maintenance Tasks" on page 97 and "Weekly Maintenance Tasks" on page 103 provide detailed procedures for the tasks summarized in the quick maintenance checklist.
- "Database Maintenance Tasks" on page 105 describes how to purge old records from the database.
- "Infrequent Maintenance Tasks" on page 111 describes some other tasks that you perform only when needed.

Quick Maintenance Checklist

Table 4-1 summarizes the daily and weekly maintenance tasks, which are covered in more detail later in the chapter.

Table 4-1 *Summary of daily and weekly maintenance tasks*

Step	Action/Command	Expected Results	Resolving Problems or Getting More Information
Daily:			
1	Stop logical log backup and replace tapes.		See page 97.

2	Check ReadiVoice processes:
	racheck
		The following processes should be running:
		`odproc` `csc`
		`megasock` `mplex`
		`sockcap` `instore`
		`vbootp`
		`sua` (IP systems only)
			Wait for all conferences to end. Run **rastop** and then **rastart**. See page 111.
			You must be root.

3	Check Informix dbspace: **onstat -d**	At least 50,000 free pages in `/dev/info/dbspace1`.	Call Polycom Global Services.
4	Check Informix users: **onstat -u**	At least four `informix`, one `web`, and three `root` users are active.	Wait for all conferences to end. Run **rastop**, reboot the Sun, and then run **rastart**. See page 111 and page 112. You must be root.
5	Check Informix logging: **onstat -l**	All logical logs are flagged `F------` or `U-B----` except the current log file, which is flagged `U---C-L`.	Start logical log backup with **ontape -c**. See page 96 and page 102.
6	Restart logical log backup and check status of its tape drive: **ontape -c** **mt -f /dev/rmt/1 status**	Device busy.	Insert tape, restart `ontape`, or troubleshoot drive. See page 102.
7	Check status of archive tape drive: **mt -f /dev/rmt/0 status**	Device at load point.	Insert tape or troubleshoot drive. See page 102.
Weekly:			
8	Check Solaris disk space: **df -k**	All disk partitions at less than 90% of capacity.	Remove old, unneeded files. See page 103.
9	Check for defunct processes: **ps -ef | grep '<defunct>' | grep -v grep**	No defunct processes listed.	Kill defunct processes. See page 104.

Understanding Informix Backup

The ReadiVoice system actually uses two Informix databases. One, `cdrcnow`, consists of all the CDR (call detail record) tables. The other, `cnow`, contains everything else, including your subscriber data. When we refer to the Informix database here, we mean the entire Informix Dynamic Server dbspace, including both of these databases.

Informix backup involves two distinct processes:

- Periodically creating a tape backup (called an *archive*) of the database.
- Continually logging all database transactions to disk and, optionally, backing up these transaction logs (called *logical logs*) to tape.

All ReadiVoice CACS servers have two 4mm tape drives, so one can be dedicated to each of these processes.

Caution! The /usr/informix/onconfig.conferencenow configuration file includes parameters specifying the archive tape device (TAPEDEV) and log tape device (LTAPEDEV). By default, both are set to /dev/null. To enable archives and logical log backups, you must change these to the correct tape drive. Typically, /dev/rmt/0 is used for archives and /dev/rmt/1 is used to continually back up logical logs.

The sections below provide an overview of each process and some guidelines for setting up your backup procedures. For more detailed information, consult the *IBM Informix Backup and Restore Guide* for Informix Dynamic Server 9.3 and other Informix documentation or call your Polycom Global Services representative.

IBM provides two data recovery systems with Informix Dynamic Server, the ON-Bar system and the `ontape` utility. Although they perform similar functions, they're not compatible with each other or interchangeable. For more information, see the *IBM Informix Backup and Restore Guide*. This manual discusses only the `ontape` utility.

At the time that this is being written, all IBM Informix documentation is available in PDF form at:

http://www-3.ibm.com/software/data/informix/pubs/library/

Archives

Archiving creates a complete (level 0) or incremental (level 1 or 2) backup of the Informix database on tape.

We don't recommend incremental archives. The ReadiVoice database is typically small enough that a level 0 archive goes quickly (under an hour), and it offers greater security and a much simpler restore process. This manual assumes that all archives are level 0.

In the event of database corruption, a crash, or catastrophic disk failure, you can restore the database from the most recent archive tape to the state it was in when the archive was created. Logical logs (see next section) then enable you to recover the transactions that took place after the most recent archive.

How often should you archive your database? It depends on the size and activity level of your system, whether you have the optional disk mirroring feature, and your comfort level with regard to risk.

We recommend, for most circumstances, nightly archives. The command **`ontape -s -L 0`** creates a full level 0 archive on the archive tape device. The process prompts for user input, so you must deal with this in the `crontab` file if you want to schedule the task.

Each morning, remove the tape, label it, and replace it with the next one. Use ten tapes (or fourteen if you operate on weekends) in rotation. Consider rotating the oldest five (or seven) to an off-site facility for protection against an extreme disaster at your site.

Caution! Validate your archives and, from time to time, test your ability to restore from them. Tapes wear out over time. Label them with the date of first use and replace them every six months.

Combining the most recent nightly archive with the subsequent logical logs (on the logical log backup tape and/or on disk), you have a very high probability of recovering all database information right up to the point of failure. Even if the logical logs on disk are lost and the logical log backup tape fails, you lose only one day's data. If an archive tape fails, you can still recover all data by using the previous day's archive tape together with the logical logs.

The procedures in "Daily Maintenance Tasks" on page 97 assume that you're archiving the database nightly.

Logical Logs

Informix records all database transactions in its logical log files as soon as they're complete. By default, your system is set up to use thirty log files of 1 Mbyte each. You can change the number and size of the log files in `/usr/informix/onconfig.conferencenow`. The logical logs provide a complete record of all database activity.

When a log is full, Informix starts writing to the next one. If all logs are full, Informix halts. Therefore, the logical logs must be backed up to tape (or to `/dev/null`) and marked as available again on an ongoing basis.

If your database crashes, you can restore it to the point of the last archive using the most recent archive tape. Then, you can recover all subsequent transactions using the logical log tape together with the log file on disk that was being used at the time of the crash. This recovers all data up to and including the last completed transaction.

Even in the event of a catastrophic disk failure, the most recent archive tape and current logical log tape let you recover all the transactions up to the point where the last logical log file was written to tape. You never lose more than the transactions in the logical log file being used at the time of the crash (assuming no tape failure).

You can adjust the size of the logical log files so that they fill up and are written to tape at a comfortable interval. For example, if it takes an average of eight hours for each logical log to fill up, you may want to reduce their size so that you don't risk losing so much data in the event of a disk failure.

We strongly recommend backing up logical logs continually to tape. If you don't, and the logical logs on disk are unrecoverable due to a disk failure, then you lose all transactions since the last archive. Since the log files *must* be sent either to tape or to `/dev/null`, backing them up to tape costs you only the price of the tapes and the time needed to change tapes and check the status of the process periodically.

We recommend using at least three tapes in rotation for logical log backup. Change tapes on at least a weekly basis and more often if necessary. Consider rotating the oldest tape(s) to off-site storage for extreme disaster recovery.

"Backing Up Logical Logs" describes how to set up continual tape backup of the logical logs. The procedures in "Daily Maintenance Tasks" on page 97 assume that you're backing up logical logs to tape.

Caution! Validate your logical log backups and, from time to time, test your ability to restore from them. Don't let the logical log backup tape become full. If it does, Informix stops backing up logical logs, and when all logical logs are full, the database halts. Tapes wear out over time. Label them with the date of first use and replace them every six months.

Backing Up Logical Logs

This procedure assumes the CACS is easily accessible. If it's not, you can:

- Use `ontape -a` to perform a one-time backup of all the full logical logs until you can get to the site to restart continual backup.
- Use a wrapper script to run `ontape -c` from a `nohup` process, directing output to a log file. If you do this, be sure you check the log file *frequently*.

Caution! Informix Dynamic Server must be configured properly to back up logical log files to tape. If you choose not to back up logical logs to tape, they must be directed to /dev/null to prevent Informix from halting when all the log files become full. Before proceeding, read "Understanding Informix Backup" on page 93.

To start logical log backup:

1. Insert a tape into the drive designated for logical log backups (we assume here that it's `/dev/rmt/1`).
2. Log into the CACS server as root, open an XTerm window, and become user informix by entering **`su - informix`**.
3. Type **`ontape -c`** and press ENTER.

   Informix responds:

   ```
   Performing continuous backup of logical logs.
   Please mount tape 1 on /dev/rmt/1 and press Return to continue.
   ```
4. Press ENTER.

   Informix backs up the currently full log files and marks them as once again available for logging. As long as the process continues to run, it backs up each log as it becomes full and marks it as available again.

   You can minimize this XTerm window if you want, but don't quit. If you do, the ontape process stops, and your logical logs are no longer being backed up.
5. In a new XTerm window, use the `onstat` command (see "Checking Informix Dynamic Server" on page 99) to verify that the system is operating properly.

Caution! Whenever you reboot the Sun, be sure to restart logical log backup. Insert a new tape when you restart if you don't want the logical logs already backed up to the existing tape to be overwritten. Don't let the logical log backup tape become full! If it does, Informix stops backing up logical logs, and when all logical log files are full, the database halts.

Daily Maintenance Tasks

This section describes tasks you should perform daily, assuming that you archive nightly and back up logical logs to tape. Perform them first thing in the morning or any time after the archive process is complete.

Stopping Logical Log Backup and Replacing Tapes

1. In the XTerm window running `ontape -c`, press CTRL+C.

   The logical log backup process stops.

   Leave this XTerm window open (or running minimized) so that later you can restart logical log backup in it. See "Restarting Backup and Verifying Tape Drive Status" on page 102.

2. Eject and replace the archive tape and, if necessary, the logical log tape.

Caution! Never try to unload a tape cartridge while the tape is in motion as indicated by the LED. Doing so can permanently damage the tape.

Checking the ReadiVoice Processes

The ReadiVoice processes are:

`odproc` — main ReadiVoice CACS process

`csc` — conference state controller

`megasock` — manages communications

`mplex` — runs Operator application

`sockcap` — captures and records log files

`instore` — collects and stores health and status information

`vbootp` — bootstraps the cards in a bridge

`sua` — call control (IP systems only)

Caution! ReadiVoice behavior is unpredictable unless all required processes are running.

To check the ReadiVoice processes:

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Change directories to `/rahome` and enter **`./racheck`** for a brief summary, or **`./check`** more detail.

   The system should show the required ReadiVoice processes running, as shown in Figure 4-1. Additional processes may also be listed.

   You can also check the processes using the ReadiVoice Administration interface. See "Checking System Status" on page 71.
3. If all required processes aren't running, follow the procedure in "Stopping and Restarting ReadiVoice Processes" on page 111.

Figure 4-1 *Checking the ReadiVoice processes*

```
SunOS 5.8

login: cnow
Password:
Last login: Thu Feb 17 16:23:05 from :0
Sun Microsystems Inc.   SunOS 5.8
$ su root
Password:
# cd /rahome
# ./racheck
   330 ?        0:01 csc
   230 ?        0:03 vbootp
   677 ?        0:01 mplex
   674 ?        0:13 odproc
   340 ?        0:00 megasock
   345 ?        0:00 instore
   669 ?        0:00 sockcap
#
```

Checking Informix Dynamic Server

1. In the same XTerm window in which you're logged in as root, switch user to informix and enter **tcsh** to set the Informix environment.
2. Check the database space by entering **onstat -d**.

   Informix displays information about the disk space that it controls. Figure 4-2 shows an example.
3. If /dev/info/dbspace1 has fewer than 50,000 free pages, call your Polycom Global Services representative.

Figure 4-2 *Checking the Informix database space*

```
su informix
tcsh
ragnar </rahome> cnow> onstat -d

Informix Dynamic Server Version 9.30.UC1   -- On-Line -- Up 12 days 19:27:51 --
34960 Kbytes

Dbspaces
address  number   flags    fchunk   nchunks  flags    owner    name
ba92150  1        1        1        1        N        informix rootdbs
ba92790  2        1        2        1        N        informix dbspace1
 2 active, 2047 maximum

Chunks
address  chk/dbs offset   size     free     bpages   flags pathname
ba92210  1    1   0        525183   403104            PO-   /dev/info/rootdbs
ba926b0  2    2   0        1024000  821333            PO-   /dev/info/dbspace1
 2 active, 2047 maximum

ragnar </rahome> cnow>
```

4. Check the Informix users by entering **onstat -u**.

   Informix displays information about the users logged into the database. With the database online and the ReadiVoice processes running, the list should include at least four informix entries, three root entries, and one web entry. Figure 4-3 shows an example.
5. If the required users aren't listed, stop the ReadiVoice processes, reboot the Sun, and restart the ReadiVoice processes. See "Stopping and Restarting ReadiVoice Processes" on page 111.

Figure 4-3 *Checking Informix users*

```
ragnar </rahome> cnow> onstat -u

Informix Dynamic Server Version 9.30.UC1 -- On-Line -- Up 12 days 19:29:52 --
34960 Kbytes

Userthreads
address  flags   sessid   user     tty      wait     tout locks nreads   nwrites
ba94018  ---P--D 1        informix -        0        0    0     14       4209
ba944cc  ---P--F 0        informix -        0        0    0     0        3577
ba94980  ---P--F 0        informix -        0        0    0     0        1619
ba94e34  ---P--F 0        informix -        0        0    0     0        0
ba973d4  Y--P--- 1172     web      -        bbddf18  0    1     0        0
ba97888  Y--P--- 1074     root     4        bb91e30  0    1     0        271
ba97d3c  ---P--D 13       informix -        0        0    0     0        0
ba981f0  Y--P--- 1075     root     4        bca7b28  0    1     0        15
ba986a4  Y--P--- 1073     root     4        bb91760  0    1     0        0
ba98b58  Y--P--- 1183     patrol   -        bbad500  0    1     0        9
 17 active, 128 total, 25 maximum concurrent

ragnar </rahome> cnow>
```

6 Check logging by entering **`onstat -l`**.

Informix displays information about physical and logical logging of the database. Figure 4-4 shows an example.

In the flags column, logical logs that have not been used (if any) are marked `F------`. Logs that have been used and backed up, and therefore are available for reuse, are marked `U-B----`. The log to which Informix is currently writing is marked `U---C-L`. Logical logs that are full, but have not been backed up are marked `U------`.

Caution! If more than one entry is marked U------, the logical logs aren't being backed up. Once all the log files reach this condition, Informix stops! Reread "Logical Logs" on page 94 if necessary and be sure you're set up properly for logical log backup (or are redirecting the log files to /dev/null). Then, restart logical log backup as described in "Restarting Backup and Verifying Tape Drive Status" on page 102.

Figure 4-4 Checking logical logging

```
ragnar </rahome> cnow> onstat -l

Informix Dynamic Server Version 9.30.UC1 -- On-Line -- Up 21 days 10:14:39 --
34944 Kbytes

Physical Logging
Buffer bufused  bufsize  numpages numwrits pages/io
  P-1  0        16       10436    923      11.31
      phybegin physize  phypos   phyused  %used
      10003f   500      441      0        0.00

Logical Logging
Buffer bufused  bufsize  numrecs  numpages numwrits recs/pages pages/io
  L-3  0        16       81863    15255    10598    5.4        1.4
        Subsystem    numrecs  Log Space used
        OLDRSAM      81863    10880128

address  number   flags    uniqid   begin        size     used    %used
a58d5b8  1        U-B----  91       100233        500      500    100.00
a58d5d4  2        U-B----  92       100427        500      500    100.00
a58d5f0  3        U-B----  93       10061b        500      500    100.00
a58d60c  4        U-B----  94       10080f        500      500    100.00
a58d628  5        U-B----  95       100a03        500      500    100.00
a58d644  6        U-B----  96       100bf7        500      500    100.00
a58d660  7        U-B----  97       100deb        500      500    100.00
a58d67c  8        U-B----  98       100fdf        500      500    100.00
a58d698  9        U-B----  99       1011d3        500      500    100.00
a58d6b4  10       U-B----  100      1013c7        500      500    100.00
a58d6d0  11       U-B----  101      1015bb        500      500    100.00
a58d6ec  12       U-B----  102      1017af        500      500    100.00
a58d708  13       U-B----  103      1019a3        500      500    100.00
a58d724  14       U-B----  104      101b97        500      500    100.00
a58d740  15       U-B----  105      101d8b        500      500    100.00
a58d75c  16       U---C-L  106      101f7f        500      265     53.00
a58d778  17       U-B----  77       102173        500      500    100.00
a58d794  18       U-B----  78       102367        500      500    100.00
a58d7b0  19       U-B----  79       10255b        500      500    100.00
a58d7cc  20       U-B----  80       10274f        500      500    100.00
a58d7e8  21       U-B----  81       102943        500      500    100.00
a58d804  22       U-B----  82       102b37        500      500    100.00
a58d820  23       U-B----  83       102d2b        500      500    100.00
a58d83c  24       U-B----  84       102f1f        500      500    100.00
a58d858  25       U-B----  85       103113        500      500    100.00
a58d874  26       U-B----  86       103307        500      500    100.00
a58d890  27       U-B----  87       1034fb        500      500    100.00
a58d8ac  28       U-B----  88       1036ef        500      500    100.00
a58d8c8  29       U-B----  89       1038e3        500      500    100.00
a58d8e4  30       U-B----  90       103ad7        500      500    100.00
ragnar </rahome> cnow>
```

Restarting Backup and Verifying Tape Drive Status

1 In the CACS XTerm window in which you stopped `ontape` (see "Stopping Logical Log Backup and Replacing Tapes" on page 97), restart logical log backup by entering **`ontape -c`**.

   Informix responds:

```
Performing continuous backup of logical logs.
Please mount tape 1 on /dev/rmt/1 and press Return to continue.
```

2 Press ENTER.

   Informix begins backing up all the currently full log files and marks them as once again available for logging. As long as the process continues to run, it backs up each log as it becomes full and marks it as available again.

   You can minimize this XTerm window if you want, but don't quit. If you quit this XTerm window, the `ontape` process stops, and your logical logs are no longer being backed up.

3 In a new XTerm window, check the status of the logical log backup tape device by entering **`mt -f /dev/rmt/1 status`**.

   The system should indicate that the tape drive is in use, as shown in Figure 4-5.

Figure 4-5 *Checking the status of the logical log tape drive*

```
ragnar </rahome> informix> mt -f /dev/rmt/1 status
/dev/rmt/1: Device busy
ragnar </rahome> informix>
```

4 Check the archive tape device by entering **`mt -f /dev/rmt/0 status`**.

   The system should indicate that the tape drive is at the load point, as shown in Figure 4-6.

Figure 4-6 *Checking the status of the archive tape drive*

```
ragnar </rahome> informix> mt -f /dev/rmt/0 status
HP DDS-3 4MM DAT tape drive:
   sense key(0x0)= No Additional Sense   residual= 0   retries= 0
   file no= 0   block no= 0
ragnar </rahome> informix>
```

This concludes the routine daily maintenance. You may want to check logical logging status again later (by entering **`onstat -l`**), after Informix has had time to fill a logical log file, to confirm that the logs are being backed up.

Weekly Maintenance Tasks

This section describes procedures that you should perform at least weekly and perhaps more often for a large and busy system.

Checking Solaris Disk Space

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Check disk space by entering **`df -k`**.

   The system displays file system statistics, including the percentage of capacity used for each partition. In the example shown in Figure 4-7, for instance, `/usr` is at 92% of capacity and requires attention.

Figure 4-7 *Checking disk space*

```
# df -k
Filesystem            kbytes    used    avail  capacity  Mounted on
/proc                      0       0        0     0%     /proc
/dev/dsk/c0t0d0s0      48023   31869    11352    74%     /
/dev/dsk/c0t0d0s5    1015695  875958    78796    92%     /usr
fd                         0       0        0     0%     /dev/fd
/dev/dsk/c0t0d0s6      92231   11219    71789    14%     /var
/dev/dsk/c0t1d0s5      92231    7707    75301    10%     /opt
/dev/dsk/c0t1d0s0     492351  246332   196784    56%     /rahome
/dev/dsk/c0t1d0s1     380311  133557   208723    40%     /web
swap                  459800  158808   300992    35%     /tmp
#
```

   You can also check the disk space from an operator/maintenance station using the ReadiVoice Administration interface. See "Checking System Status" on page 71.

3. If any partition other than root (/) is at or above 90% of capacity, you should remove old, unneeded files to free up space. Call your Polycom Global Services representative if you need help in determining what you can safely remove.

Caution! If you run out of space on any disk partition, Solaris ceases to operate!

Checking for Defunct Processes

1 As root, enter **ps -ef | grep '<defunct>' | grep -v grep**.

The system lists any defunct processes (processes that have no parent). If there are none, the prompt simply reappears. The example in Figure 4-8 shows three defunct processes.

Figure 4-8 *Checking for defunct processes*

```
# ps -ef|grep '<defunct>'|grep -v grep
   jhaug   123   122  0                   0:00 <defunct>
    root 14919   377  0                   0:00 <defunct>
   jhaug  5807  5806  0                   0:01 <defunct>
#
```

2 If there are defunct processes, note their process IDs (PIDs). The PID is the first number after the process owner's name. For instance, in the example in Figure 4-8, the PID of the first defunct process is 123.

3 Kill each defunct process by entering **kill *PID***, where ***PID*** is the PID number for the process.

4 Enter **ps -ef | grep '<defunct>' | grep -v grep** again to confirm that the processes were killed.

5 If the basic kill command failed to kill a process, enter **kill -9 *PID***. Then check again to confirm that it was killed.

Caution! Be very careful when using the kill command and especially when using kill -9. Be certain that you enter the correct PID. Never kill an sqlturbo process or you may bring down the database.

6 Restart any processes that you had to kill, but that should be running.

Database Maintenance Tasks

Without proper maintenance, your ReadiVoice databases will grow ever larger, slowing the system. It's imperative that you routinely remove (purge) old records. Polycom provides several scripts you can use for this purpose.

You can run these scripts manually, but you'll probably want to schedule them to run automatically, using the cron task scheduler. The sections that follow describe these scripts. Read through these sections to determine which scripts you need to run and how to run and schedule them.

Caution! Run only the scripts described in this section or elsewhere in this manual. Other scripts in the file system must not be run without express instructions from your Polycom Global Services representative.

Purging Outdated Call Detail Records (CDRs)

Your ReadiVoice installation includes a script, `purgeCDR`, that lets you easily remove old CDR data from your database. You can run the `purgeCDR` script:

- Manually from the command line (called interactive mode).
- Automatically at periodic intervals, using the cron task scheduler.

Either way, the `purgeCDR` script:

- Purges from the database all the CDRs that are older than a cutoff date or a number of days that you specify. By default, it purges only conferences with the processed flag set to 1, but you can tell it to ignore the flag.
- By default, writes purged records to a text file named *YYYYMMDDhhmmss*`cdr.log` (where *YYYYMMDDhhmmss* is the run date of the purge) and then archives the file as *YYYYMMDDhhmmss*`cdr.log.gz`. The archives are stored in `/rahome/cdr/archives`. Archiving can be turned off.

When you run the `purgeCDR` script from the command line, you can view a list of conferences to be purged without actually purging them.

CDR Purge Script Command Syntax and Examples

The `purgeCDR` script resides in `/rahome/cdr`. When you run `purgeCDR`, it calls another script, `purgeCDR.pl`, in the same directory. This second script performs the actual purge process; `purgeCDR` can monitor the CPU usage and halt `purgeCDR.pl` as needed in order to minimize the impact on your system.

Caution! We strongly discourage running purgeCDR.pl directly on a live conferencing system. Doing so may impede your ReadiVoice conferencing services.

The `purgeCDR` command syntax is:

```
purgeCDR [-h] [-d mm/dd/yyyy] [-n numOfDays] [-i] [-u] [-m maxCpuUsage] [-rt numOfMinutes]
    [-cs numOfSubscribers] [-v] [-ip] [-na]
```

Table 4-2 describes the options (command-line arguments).

Table 4-2 *purgeCDR command options*

Option	Description
`-h`	Display help.
`-d`	Cutoff date. CDRs of conferences that ended before this date are purged. Must be in *mm/dd/yyyy* format, where *mm* is month, *dd* is day, and *yyyy* is year. This option is only useful for running the script manually (interactive mode). Either it or the next must be specified.
`-n`	Cutoff number of days. CDRs of conferences that ended more than this many days ago are purged. May be any integer. Use this option to run the script automatically at scheduled intervals. Either it or the `-d` (date) option must be specified.
`-i`	Run in interactive mode (manually at the command line as a one-time purge). The script prompts you to confirm before purging the records. This option must be specified in order to use the `-u` option.
`-u`	Updates database statistics (indexes) upon completion. The script must be run by user cnow to use this option.
`-m`	Maximum CPU usage. The purge process halts temporarily if CPU usage (in percent) by other processes exceeds this value. Valid values are 0 to 100. Doesn't check CPU usage if not specified. When running purgeCDR on a live conferencing system, we strongly recommend using this option to minimize the impact on your system.
`-rt`	Maximum run time in minutes. Default is 120.
`-cs`	Number of conferences to delete per batch. Default is 4000. Valid values are 50 to 4000.
`-v`	Display a list of conferences to be purged without purging them. If you use this option with cutoff parameters that return a very large number of records (>50,000), the script may terminate with an "out of memory" error.
`-ip`	Ignore the `processed` flag in the `cdr_post_conf` table. If this option isn't specified, the script deletes only conferences with the `processed` flag set to 1.
`-na`	No archive. Specify this option to skip writing the records to a file before purging.

Both the cutoff date and number of days options use midnight as the boundary. If you specify a cutoff date of 08/21/2003, the last conference purged will be one that ended at 11:59:59 on 08/20/2003. If you specify the number of days as 30, and 30 days ago was 08/14/2003, the last conference purged will be one that ended at 11:59:59 on 08/13/2003.

Table 4-3 gives examples of the command with various options and values and describes the result of running the command.

Table 4-3 *purgeCDR examples*

Command	Description
`purgeCDR -d 07/29/2005 -v`	Displays a list of all conferences that ended before July 29, 2005, but does *not* purge them. Since -ip isn't specified, it lists only those with the processed flag set.
`purgeCDR -d 07/29/2005 -i -rt 60`	Issued at command line (interactive mode). Purges all CDR records associated with conferences flagged as processed that ended before July 29, 2005. Runs for up to 60 minutes.
`purgeCDR -n 5 -i -m 20 -ip -na`	Issued at command line (interactive mode). Purges all CDR records associated with conferences that ended more than five (5) days ago. Doesn't check the processed flag or create an archive file. Process halts temporarily if CPU usage by other processes exceeds 20 percent.
`purgeCDR -n 5 -m 40 -cs 1000 -ip -u`	Suitable for scheduling in cron (non-interactive mode). Purges all CDR records associated with conferences that ended more than five days ago. Process halts temporarily if CPU usage by other processes exceeds 40 percent. Deletes the conferences 1000 at a time. Doesn't check the processed flag. Updates statistics when finished.

Running a One-Time Purge of CDRs

To manually purge CDRs from the command line:

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Change to the `/rahome/cdr` directory.
3. Type the `purgeCDR` command with the options you want and press ENTER.

   To see which CDRs will be purged, but not purge them, use the `-v` (view) option.
4. At the prompt to continue, type **y** and press ENTER. Then wait for the process to complete.

   If you're purging records (not using the `-v` option), the script removes the out-of-date records from the database, writes them to a text file named *YYYYMMDDhhmmss*`cdr.log` (where *YYYYMMDDhhmmss* is the run date of the purge) and then archives the file as *YYYYMMDDhhmmss*`cdr.log.gz`.
5. When the script has finished, verify that the records were deleted and that the backup file was created.

   The backup file is in `/rahome/cdr/archives`.

Scheduling Automatic Purges of CDR Records

To schedule `purgeCDR` to run periodically and automatically in the cron task scheduler:

1. Decide how many days you want to retain CDR records in the database. Use this number as the argument of the `-n` option. If you're not flagging conferences as processed, specify the `-ip` option.
2. Decide what the maximum CPU usage should be before the `purgeCDR` script suspends operation. Use this number as the argument of the `-m` option.
3. For your records, write the command on the line below as you want it to run. For instance, if you want to purge records more than 30 days old, suspend the process when CPU usage exceeds 40%, and ignore the processed flag, write the command as **`purgeCDR -n 30 -m 40 -ip`**.

   ____

   D*on't* use the `-d`, `-i`, or `-v` options, which aren't suitable for automatic scheduling.
4. Decide how often you want to run the purge process. We recommend a frequency from weekly to daily, depending on your ReadiVoice system's conferencing and database usage.
5. In the root user's `crontab` file, schedule the command to run at the time and frequency you want. See the Solaris documentation or the crontab and cron man pages for information on scheduling tasks with cron.

Restoring Purged CDR Records

Should the need arise, you can restore purged CDRs to your database using the `loadtable.pl` script referred to in "Restoring from a Manual Backup Tape" on page 116.

To restore all the CDRs from an archive file created by `purgeCDR`:

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Change directories to `/rahome` and enter **`./rastop`**.
3. Set the Informix environment by entering **`tcsh`**.
4. Change directories to `/rahome/cdr/archives` and find the archive file containing the records you want to restore (its name is *`YYYYMMDDhhmmss`*`cdr.log.gz`, where *`YYYYMMDDhhmmss`* is the run date of the purge).
5. Unzip the archive, extracting *`YYYYMMDDhhmmss`*`cdr.log`.

6 Restore the data from the CDR backup file to the `cdrcnow` database by entering (all on one line):

```
/rahome/dbcdr/loadtable.pl DB=cdrcnow <
      /rahome/cdr/archives/YYYYMMDDhhmmsscdr.log
```

Replace ***YYYYMMDDhhmmss*`cdr.log`** with the name of the backup file.

7 Change directories to `/rahome` and enter **`./rastart`**.

8 Log into the Administration interface and use the **CDR Data** pages to verify that the records you wanted to restore are there (see "Viewing Conferencing Information" on page 72).

Purging Subscriber Records Flagged for Deletion

When you change a subscriber's status to deleted, either in the Provisioning interface or with a stored procedure call, the ReadiVoice system doesn't actually delete the subscriber from the database; it merely flags the record for deletion. To actually delete such records from the database, you must run the `subDelete.pl` script.

Caution! The subDelete.pl script ***immediately*** and ***permanently*** deletes all subscriber records whose status is set to **2** (deleted) in the database. This script does ***not*** create a backup of the deleted records. ***Be sure you really want to delete these subscribers.*** Be sure your provisioning procedures are appropriate (that is, the guidelines for setting a subscriber's status to deleted take into account the use of this script).

To *permanently* and *irrevocably* delete all subscriber records flagged for deletion:

1 Open a Telnet session to the CACS as user cnow.

2 Change to the `/rahome/database/scripts` directory.

3 Type **`./subDelete.pl`** and press ENTER.

All subscriber records flagged for deletion are permanently deleted from the database.

Caution! You can schedule this script as a cron task. But, remember that purged subscriber records, unlike purged CDRs, can't be recovered. Therefore, approach the automatic deletion of subscriber records with caution.

Purging the CacsEventUpdate Table

If you use the Provisioning Stored Procedure Interface (PSPI) for provisioning, you need to purge the `CacsEventUpdate` table from time to time to keep it from growing too large.

Several stored procedures don't load the new data into call router memory immediately. Instead, they put the new data into the `CacsEventUpdate` table; the CACS then updates the call router with the data from the temporary table every ten seconds.

For more information about PSPI, see the *ReadiVoice PSPI Reference*.

If you use stored procedure calls (SPCs) that write records to this table, then you must purge those records regularly.

Your ReadiVoice installation includes a script, `cleanCacsEvtUpdate.pl`, that purges the processed records from the `CacsEventUpdate` table. You can use the cron task scheduler to run this script periodically.

Caution! Using this script improperly may impede your ReadiVoice conferencing services. Your Polycom Global Services representative can help you determine the best frequency, time, and run time parameter for this script.

The `cleanCacsEvtUpdate.pl` script resides in `/rahome/database/scripts`. The syntax is:

`cleanCacsEvtUpdate.pl [-rt nn]`

The `-rt` parameter lets you specify a run time, *nn*, in minutes. If you omit it, the run time defaults to 10 minutes. The script terminates at the end of the run time, even if it hasn't purged all processed records from the table.

Purging the AccOptChanges Table

If your system has the Account Options Updates feature enabled (see "Enabling Account Options Updating" on page 128), you need to purge the `AccOptChanges` table from time to time to keep it from growing too large.

If the Account Options Updates feature is enabled, when subscribers change their account options (using the account options menu in the call flow), the ReadiVoice system records those changes in the `AccOptChanges` table (in addition to updating the subscriber records).

You can use the `GetAccOptChanges` stored procedure to retrieve these account options update records and update your customer database. The stored procedure flags the records you retrieve as processed.

For more information about this stored procedure, see the *ReadiVoice PSPI Reference*.

Your ReadiVoice installation includes a script, `cleanAccOptChanges.pl`, that purges the processed records from the `AccOptChanges` table. You can use the cron task scheduler to run this script periodically.

The `cleanAccOptChanges.pl` script resides in `/rahome/database/scripts`. The syntax is:

```
cleanAccOptChanges.pl [-rt nn]
```

The `-rt` parameter lets you specify a run time, *nn*, in minutes. If you omit it, the run time defaults to 10 minutes. The script terminates at the end of the run time, even if it hasn't purged all the processed records from the table.

The `cleanAccOptChanges.pl` script suspends the purge process temporarily if CPU usage by other processes exceeds 30%. Nevertheless, it's best to schedule execution for times of low load. Your Polycom Global Services representative can help you choose the best frequency, time, and run time parameter for this script.

Infrequent Maintenance Tasks

This section describes procedures that you perform only as needed, not on a regular basis.

Stopping and Restarting ReadiVoice Processes

The ReadiVoice processes are:

`odproc` — main ReadiVoice CACS process

`csc` — conference state controller

`megasock` — manages communications

`mplex` — runs Operator application

`sockcap` — captures and records log files

`instore` — collects and stores health and status information

`vbootp` — bootstraps the cards in a bridge

`sua` — call control (IP systems only)

ReadiVoice behavior is unpredictable unless all required processes are running. Under certain circumstances, you should stop all ReadiVoice processes and then restart them. These include:

- If a ReadiVoice process isn't running.
- If you need to change the date and time on the Sun server.

  In this case, you should also reboot the Sun. See "Rebooting the Sun Server" on page 112.
- If you change the system configuration settings for routing, confidence, or hidden number use.

Caution! Stopping and restarting the ReadiVoice processes affects current users as described below.

It's best to stop and restart ReadiVoice processes only when there are no active conferences. In an emergency, however, you may have to do this while conferences are running. If so, keep the following in mind:

- Anyone using the Operator or Moderator interface is isolated from the conferencing system for the time it takes to stop and restart (about one minute).
- New calls can't be handled until the core processes have restarted. Depending on your system configuration, callers hear either a busy signal or a message stating that the system isn't available and asking them to call back.
- Existing conferences continue to operate normally. The participants are unaffected unless they request an operator or try to use the Moderator interface during the stop/restart operation.

To stop and restart the ReadiVoice processes:

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Change directories to `/rahome`.
3. Enter **`./rastop`**. If the prompt doesn't reappear, press ENTER.
4. Enter **`./rastart`**. If the prompt doesn't reappear, press ENTER.
5. To verify that the required processes are running, enter **`./racheck`**.

Rebooting the Sun Server

Under certain circumstances, you should reboot the server. These include:

- If you change the date and time on the Sun server.
- If the system exhibits erratic or unpredictable behavior.

Caution! Rebooting the Sun server affects current ReadiVoice users as described in the previous section and below.

Naturally, rebooting the server also includes stopping and restarting the ReadiVoice processes. If conferences are running, the same considerations apply. But, the entire process takes longer, perhaps five to ten minutes. In addition, the bridges don't have access to files on the server during the reboot. This may affect existing conferences if, for example, sound files not already cached on the bridge are needed.

To reboot the Sun server:

1 As root, stop the ReadiVoice processes as described above.

2 To perform a warm reboot in 30 seconds, enter **`shutdown -g30 -i6 -y`**.

   Solaris begins shutting down in 30 seconds without prompting for verification. The entire reboot process takes several minutes.

   Omit **`-y`** to get a confirmation prompt. Change the value after **`-g`** to change the delay. See the man page for the shutdown command for more options.

3 Once Solaris has finished rebooting, restart the ReadiVoice processes as described above.

Rebooting a Bridge

Under certain circumstances and with the guidance of Polycom Global Services, you may need to reboot a bridge.

A cold boot means that you turn off power to the entire bridge and then turn it back on.

A warm boot means that power to the bridge is always on while you individually re-start various components. You can re-start many components either through the software or through reset switches on the components themselves.

Due to the number of cards, it's impractical to perform a warm boot of an entire bridge. Individual cards can be warm booted, which might be preferable if the problem and its resolution can be localized. Rebooting individual cards permits the remaining cards to continue to service conferences and share resources.

You should have Polycom Global Services reboot individual VCE or HMod cards. Global Services representatives can do this remotely. They can also determine whether a card is in use and can set the card as least-preferred for all new conferences and for resource sharing between cards. This allows the card to be rebooted without affecting any conferences on the bridge.

To reboot (cold boot) a bridge:

1 On the System Administration interface's **Bridges** page, change the status of the bridge to *Busyout* (see "Modifying an Existing Bridge" on page 20).

2 Wait for all conferencing activity to end.

Caution! Rebooting a bridge terminates all conferencing activity! Reboot only when there are no active conferences on the bridge.

3 Using the power switch on the rear of the bridge cabinet, turn off power to the bridge.

The VCE/HMod cards in the bridge must be able to access files on the server in order to boot and to initiate various processes. If you're also rebooting the server, be sure it has finished booting before restoring power to the bridge.

4. Wait at least fifteen seconds. Then, turn on the bridge.

   The bridge powers up and the VCE/HMod cards boot from the server.

Changing the ReadiVoice Password

From time to time, you may want to change the ReadiVoice cnow user password. It's defined in several places, so you must change:

- The login password for cnow on the server (and on a boot server, if used).
- The password for both the cnow user and the SYSTEM (Telnet) cnow user in the ReadiVoice application (several places) and in the `vbootp.db` bridge configuration file.

Your ReadiVoice installation includes a shell script, `cnow_password_change`, that lets you easily make all the password changes for the ReadiVoice application and bridges.

Caution! Perform this procedure only during maintenance periods when there are no active conferences.

To change the cnow password:

1. Open a Telnet session to the CACS as user cnow and switch user to root (or log into the CACS as root and use an XTerm window).
2. Change directories to `/rahome/bin` and enter **`cnow_password_change`**.

   The script prompts you twice to enter the cnow user password.

3. Enter and confirm the new password.

   The script prompts you twice to enter the SYSTEM cnow user password.

4. Enter and confirm the new password. Be sure it's the same password you entered in the previous step.

   The script encrypts the new passwords and updates them in various application configuration files and processes, the database, and the `vbootp.db` file (from which the bridges will be updated).

5. If your system includes one or more boot servers, repeat the preceding steps on the boot server(s).
6. One at a time, reboot the bridges.

   Especially for InnoVox 4000 bridges, it's important to wait until a bridge finishes booting before rebooting the next bridge.

Manually Backing Up and Restoring

You may want to manually back up your ReadiVoice system at times, such as prior to an upgrade. The backed-up data lets you to revert to the previous version if there's a problem with the upgrade.

The procedure described here backs up the databases, along with the contents of the `/rahome` directory, which contains most of your configuration files.

Software upgrades automatically bring forward all database information during the upgrade process. If you need to revert to an earlier version of the ReadiVoice software, you must restore databases that match the earlier software version.

Backing Up Manually

1. Place a tape in the archive tape drive (typically, `/dev/rmt/0`).
2. In an Xterm window, as root, set the Informix environment by entering **`tcsh`**.
3. Verify that the tape is ready (see "Restarting Backup and Verifying Tape Drive Status" on page 102).
4. Change directories to `/rahome/database`.
5. Dump the ReadiVoice tables to a file by entering (on one line):

   /rahome/database/dumptable.pl DB=cnow >
   /rahome/dbdata_*mmddyyyy*.sql

   Replace ***mmddyyyy*** with the current date.
6. Dump the CDR tables to a file by entering (on one line):

   /rahome/database/dumptable.pl DB=cdrcnow >
   /rahome/cdrdata_*mmddyyyy*.sql

   Replace ***mmddyyyy*** with the current date.
7. Change to the root directory and back up the `/rahome` directory to tape by entering **`tar cvf /dev/rmt/0 /rahome`**.
8. When the backup is finished, verify the contents of the tape by entering **`tar tvf /dev/rmt/0`**.
9. Eject the tape, write-protect it, and label it with the date and the ReadiVoice version number.

Label all backups with a version number! If you downgrade to a previous version of the ReadiVoice software, you must also restore databases matching that version. The `loadtable.pl` process fails if the data being restored doesn't match the ReadiVoice version installed.

Restoring from a Manual Backup Tape

1. In an XTerm window, as root, stop the Web server process by entering **`sh /etc/rc3.d/S91rvhttpd stop`**.
2. Change directories to `/rahome` and enter **`./rastop`**.
3. Extract the backup data from tape by entering **`tar xvf /dev/rmt/0`**.

   The system restores the backup of the `/rahome` directory, including the database dump files you created there.
4. Switch user to cnow. Set the Informix environment by entering **`tcsh`**.
5. Change directories to `/rahome/database`. Then, drop the existing `cnow` database by entering **`make drop`**.
6. Create a new empty `cnow` database by entering **`make create`**.
7. Change directories to `/rahome/cdrdb`. Then, drop the existing `cdrcnow` database by entering **`make drop`**.
8. Create a new empty `cdrcnow` database by entering **`make create`**.
9. Restore the data from the dump file to the `cnow` database by entering (on one line):

   ```
   /rahome/database/loadtable.pl DB=cnow <
         /rahome/dbdata_mmddyyyy.sql
   ```

   Replace ***mmddyyyy*** with the date of the backup.
10. Restore the data from the CDR dump file to the `cdrcnow` database by entering (on one line):

    ```
    /rahome/dbcdr/loadtable.pl DB=cdrcnow <
          /rahome/cdrdata_mmddyyyy.sql
    ```

    Replace ***mmddyyyy*** with the date of the backup.

Caution! You must use the `loadtable.pl` script in `/rahome/dbcdr`, not the one in `/rahome/database`, to restore the CDR data.

11. Exit from the cnow login (returning you to the root prompt) and restart the Web server process by entering **`sh /etc/rc3.d/S91rvhttpd start`**.
12. Change directories to `/rahome` and enter **`./rastart`**.
13. To verify that the required processes are running, enter **`./racheck`**.

5

Configuring the ReadiVoice System

This chapter describes how to change your ReadiVoice system's configuration settings and which capabilities, features and options are enabled.

You can perform some of these tasks in the System Administration interface, but many of them involve editing configuration files and working at the UNIX command prompt. Before undertaking these tasks, you should be knowledgeable about Solaris UNIX, Sun servers, personal computers, software configuration, telephony, and networking.

Caution! If your ReadiVoice system is covered by warranty or a service contract, don't attempt to move, reinstall, or reconfigure the system or make any other hardware or network changes without first consulting Polycom Global Services. Doing so may void your warranty. Even in the absence of such coverage, we *strongly* urge you to contact Polycom before proceeding. We can give you a quote for assistance ranging from remotely supporting your planned change to sending a support specialist to your site to implement the change for you.

Before you proceed with any configuration change, review your installation and upgrade records and log files to verify that your desired configuration is compatible with your current configuration.

For descriptive simplicity, this chapter assumes that all components relating to SCP or SIP call control, databases, and Web servers reside on a single Sun workstation that serves as the CACS. If you have a larger system, these components, if used, may be distributed among multiple workstations.

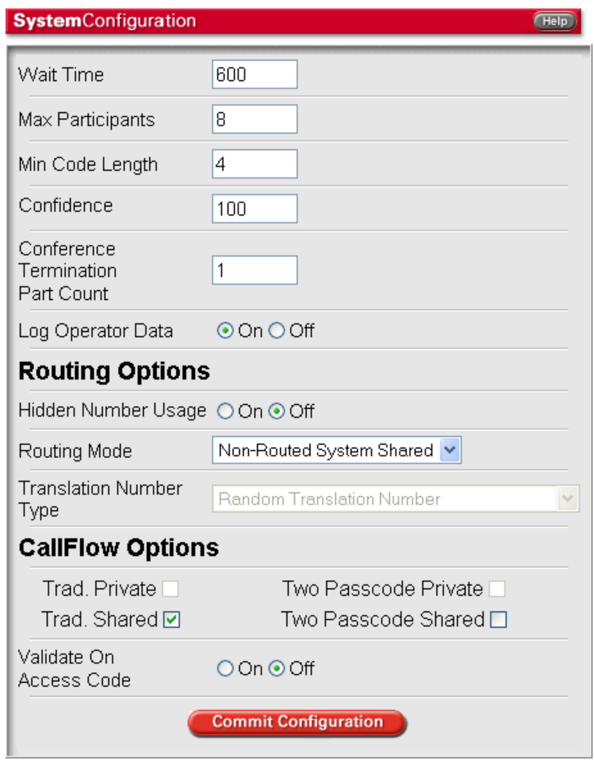
Changing System Configuration Settings

Caution! Polycom sets your system configuration options to the correct values for your installation. Before making any changes, contact your Polycom Global Services representative.

1 In the **System Administration** navigation bar, click **System Configuration**.

The **System Configuration** page appears (Figure 5-1).

Figure 5-1 ***System Configuration*** *page*



2 Make any changes to the first five settings that you want to make. The settings are:

Wait Time — Determines how long (in seconds) the first conference participant can wait on hold for the subscriber to dial in and start the conference. Additional participants may wait for less time depending on when they arrive for the conference. The default is 600 (ten minutes). The maximum value is 32,767.

Max Participants — The maximum number of participants permitted in a conference. The default is 8. The maximum value is 300.

Min Code Length — The the *minimum* number of digits permitted in an access code, subscriber password, or participant password. The default is 4. The maximum value is 20.

Confidence — the *Statistical Port Management Confidence Factor* determines how aggressively the system attempts to manage port utilization. It can be set to any integer value from 0 to 100. The default is 100, which is the most conservative setting. At this setting, the system allocates every new conference 100% of the ports that it's entitled to use (the **Max Participants** setting for the subscriber account). This guarantees that every running conference can always accommodate its entire subscribed capacity.

A lower value lets you "oversubscribe" the system's capacity on the premise that not all users will access the system simultaneously or use their entire subscription size. For best overall results, Polycom recommends a value between 85 and 95.

Before changing this setting, contact your Polycom Global Services representative for help in choosing an appropriate value. We have statistics tables, based on an average conferencing profile, for calculating an appropriate confidence level. Or, Voyant can statistically model your actual conferencing history to tune your system for maximum efficiency. This modeling typically requires about six to nine months of usage data.

Caution! Overuse of the VIP Conference option (see page 56) can have a negative impact on your system's port utilization efficiency regardless of the Confidence setting.

3 Select **On** or **Off** for **Log Operator Data**. If you turn on this option, then every night at midnight the ReadiVoice system stores operator data collected the previous day. You can view this data from the Operator Statistics page (see "Viewing Operator Information" on page 77).

4 Select **On** or **Off** for **Hidden Number Usage**. If, for internal routing purposes, your carrier maps your system's access phone numbers to different numbers, select **On** and use the Access Numbers pages to associate every access phone number to its own hidden (routing) number. Otherwise, or if this is an IP system, select **Off**.

5 Select a supported routing mode, translation number type, and call flow for your system. See page 121 for the supported configurations.

Caution! Don't change these settings without first contacting Polycom Global Services. If these settings are wrong, callers can't reach your system! Furthermore, changing your routing mode requires additional changes to both your telephone network connections and your ReadiVoice configuration.

Contact your Polycom Global Services representative if you have any questions about how calls are routed to your system or want to discuss changes to your system's call routing configuration.

If you change **Routing Mode** from **Non-Routed System Private** to **Non-Routed System Shared**, delete all existing (private) access phone numbers and make sure that the Provisioning pages have the Access Code field available (see "Setting Up the Provisioning Interface" on page 60). For ReadiVoice IP, **Routing Mode** must always be set to **Routed System**.

Routing Mode

The routing mode determines how calls are routed among bridges:

- **Non-Routed System Shared** — Each trunk group terminates on a specific bridge, so the access phone number determines which bridge receives a call. Each access phone number can be used by multiple subscribers.
- **Non-Routed System Private** — Each trunk group terminates on a specific bridge, so the access phone number determines which bridge receives a call. Each access phone number can be used by only one subscriber, uniquely identifying the subscriber.
- **Routed System** — Uses an Intelligent Network Call Routing (INCR) system, such as SS7, to tell the network how to route a call. For each number group, a routing list specifies which bridge or bridges the conferences in that group can use and the priority order of those bridges.

Translation Number Type

In a routed PSTN system, when someone dials a ReadiVoice access number, the originating switch queries the CACS, which returns a *translation number* to tell the telephone network how to route the call. In a non-routed or IP system, each bridge is provisioned with a block of dummy translation numbers.

The **Translation Number Type** setting dictates how translation numbers are assigned. The following settings are available:

- **Random Translation Number** — The system uses translation numbers from the pool of numbers assigned to a bridge. All non-routed systems must use this setting.
- **Fixed Translation Number:** The system uses a predetermined translation number for each subscriber. The translation number is stored in the `ExternalId` field of the subscriber record.

  If you select this setting, then the `ExternalId` field for each subscriber must be unique (that is, no two subscribers on this ReadiVoice system may have the same external ID). In addition, the **Trans DNIS Length** setting for the bridges (see "Maintaining Bridge Information" on page 18) must exactly equal the number of digits in the external ID.
- **Fixed Translation Number with 3-digit code:** The system also uses a predetermined translation number for each subscriber (stored in the `ExternalId` field of the subscriber record). But it appends a 3-digit number (internally-formed) to each translation number when the translation number is used by the call router. This setting is used for leg-based billing.

If you select this setting, then the `ExternalId` field for each subscriber must be unique (that is, no two subscribers on this ReadiVoice system may have the same external ID). In addition, the **Trans DNIS Length** setting for the bridges (see "Maintaining Bridge Information" on page 18) must be exactly three more than the number of digits in the external ID.

Call Flow Options

ReadiVoice has the following call flow options:

– **Traditional Private:** With this option selected, subscribers can be provisioned with private access numbers. Not available if **Routing Mode** is set to **Non-Routed System Shared**.

– **Traditional Shared:** With this option selected, subscribers can be provisioned with shared access numbers. Not available if **Routing Mode** is set to **Non-Routed System Private**.

– **Two Passcode Private:** With this option selected, subscribers can be provisioned with private access numbers and participants can be required to enter a participant password. Not available if **Routing Mode** is set to **Non-Routed System Shared**.

– **Two Passcode Shared:** With this option selected, subscribers can be provisioned with shared access numbers. Not available if **Routing Mode** is set to **Non-Routed System Private**.

Supported Call Flow Combinations

Voyant has tested and supports the following routing mode and call flow combinations:

– **Non-Routed System Shared:** Only the shared call flow options are available. Voyant has tested and supports the combinations shown in Table 5-1.

Table 5-1 *Supported call flows on non-routed shared systems*

Call Flow Options	Translation Number Type
Traditional Shared	Random
Traditional Shared + Two Password Shared	Random

– **Non-Routed System Private:** Only the private call flow options are available. Voyant has tested and supports the combinations shown in Table 5-2.

Table 5-2 *Supported call flows on non-routed private systems*

Call Flow Options	Translation Number Type
Traditional Private	Random
Two Password Private	Random

– **Routed System:** The **Traditional Shared**, **Traditional Private**, and **Two Passcode Private** call flow options are available. Voyant has tested and supports only the combinations shown in Table 5-3.

Table 5-3 *Supported call flows on routed systems*

Call Flow Options	Translation Number Type
Traditional Shared	Random
Two Password Private	Fixed Translation Number
Traditional Shared + Two Password Private	Fixed Translation Number with 3-digit code

6 Select **Yes** or **No** for the **Validate on Acc Code** option. This option isn't available if **Routing Mode** is **Non-Routed System Private**.

– **Yes:** The ReadiVoice system identifies a caller's conference using the access code only.

– **No:** The ReadiVoice system identifies a caller's conference using the DNIS/access code pair.

7 To implement your changes, click **Commit Configuration**.

The system confirms your changes and provides a link back to the **System Configuration** page. If necessary, a note reminds you that you must restart the application.

8 Click the link to return to the **System Configuration** page and verify that the settings are correct.

9 If you changed any of the routing or call flow options, you must run the `tnl_mkln` script:

a Log into the CACS as root and, in an XTerm window, enter **`tcsh`**.

b Change to the `/rahome/bin` directory.

c Run the make link script by entering **`tnl_mkln`**.

d One at a time, reboot the bridge(s).

Especially for InnoVox 4000 bridges, it's important to wait until a bridge finishes booting before rebooting the next bridge.

10 If necessary, restart the ReadiVoice application (see "Stopping and Restarting ReadiVoice Processes" on page 111).

Changing the Talker Update Frequency

Through the Operator and Moderator interfaces (and the APIs), the ReadiVoice system can report which two channels in a conference have talk slots. By default, talker information is updated every two seconds, but this interval can be lengthened to reduce the load on the system.

Caution! Perform the following procedure after operational hours. Don't reduce the interval to less than two seconds.

To change the talk slot update interval for the system:

1. Open a Telnet session to the CACS and log in as cnow (or log directly into the CACS as root and switch user to cnow).
2. Open the `.odprocrc` file for editing, and find the `[bridgeInt]` section.

   The `updateIntervalFrequency` keyword sets the talker update frequency. By default, it's set as follows:

   ```
   [bridgeInt]
   ...
      updateIntervalFrequency = 2
   ```

3. To reduce the update frequency (lengthen the interval), change the value to an integer greater than 2. Then, save the `.odprocrc` file.
4. As root, stop and restart the ReadiVoice system as described in "Stopping and Restarting ReadiVoice Processes" on page 111.

Configuring Shortened Dial-Out Call Flow

In the standard dial-out call flow, after the subscriber enters and confirms the phone number to dial, the system supports four commands for processing the call. The help menu played to the subscriber (`do_long_cmd_help.wav`) describes the options as follows:

> "After the call is answered, to connect the line into the conference, press star 1. To connect the line and continue dialing, press star 2. To disconnect the line, press star 3. To disconnect the line and continue dialing, press star 4."

To provide faster and simpler dial-outs, your system can be configured to reduce these four options to two. In the shortened dial-out configuration, the help menu (`do_short_cmd_help.wav`) describes the options as follows:

> "Once the call is answered, to place the participant into the conference, press star 1. To disconnect the participant, press star 2."

This is a system-wide configuration. If the shortened call flow is in effect, you can shorten it further by also turning off dialed number confirmation (so that the system doesn't repeat the dialed number back to the subscriber, who must press the # key to confirm).

Caution! Perform the following procedure after operational hours.

To configure your system to use the shortened dial-out call flow:

1. If you've customized or added prompt sets, review the voice prompt files used in the shortened dial-out call flow configuration and, if necessary, install replacements or customized versions of these files (see "Working with Prompt Sets" on page 166 and). The four voice prompt files are:

   ```
   do_part_disconnected.wav
   do_part_joined.wav
   do_short_cmd_help.wav
   do_short_number_prompt.wav
   ```

   Contact your Polycom Global Services representative for help with voice prompts.

2. Open a Telnet session to the CACS and log in as cnow (or log directly into the CACS as root and switch user to cnow).

3. Change to the `/rahome/bridge/scripts` directory and remove the long dial-out script by entering:

   `rm inDialSub.x`

4. Establish a symbolic link to the shortened dial-out script by entering:

   `ln -s inDialSubShort.x inDialSub.x`

5 To also turn off dialed number confirmation:

   a As user cnow, open the `ive.ini` file for editing.

   b In the `[Dialout]` section, add the following line:

   ```
   ConfirmDialNum = 0
   ```

   c Save and close the file.

6 Reboot the bridges.

   Especially for InnoVox 4000 bridges, it's important to wait until a bridge finishes booting before rebooting the next bridge.

Disabling Waiting Room Notifications

If your subscribers use the Waiting Room feature (which puts callers to a locked conference on hold until admitted by the subscriber), the ReadiVoice system notifies the subscriber when a caller enters the waiting room and periodically reminds the subscriber that callers are waiting to be admitted.

If your customers find these waiting room notification messages too intrusive, you can configure your system not to play them. In that case, however, your customers should be instructed to use the Moderator whenever a conference is locked with Waiting Room on. The Moderator interface shows that callers are waiting and facilitates the processing of those callers.

Caution! Perform the following procedure after operational hours.

To disable waiting room notifications:

1 Review the voice prompt files that mention the Waiting Room notifications feature and have replacements or customized versions recorded. The five voice prompt files are:

   ```
   cf_in_conf_subs_cmd_help.wav
   wr_cmd_help.wav
   _wr_new_caller.wav
   wr_notification_off.wav
   wr_notification_on.wav
   ```

   For more information about voice prompts, see "Working with Prompt Sets" on page 166 and Appendix A. Contact your Polycom Global Services representative for help with voice prompts.

2 As user cnow, install the replacement `.wav` files.

3 As user cnow, open the `ive.ini` file for editing and find this section near the end:

```
[MiscConfig]

   WRAnnounceDefault  =  1
```

4 Change the value after the `WRAnnounceDefault` keyword from `1` to `0`. Then, save and close the `ive.ini` file.

5 Reboot the bridges so that they read the updated `ive.ini` file and load the new `.wav` files.

Especially for InnoVox 4000 bridges, it's important to wait until a bridge finishes booting before rebooting the next bridge.

Configuring Dial-Out Billing

The dial-out billing feature (also known as unattended dial-out, or *UDO,* billing) facilitates billing when subscribers dial out to additional participants without operator assistance.

With this feature enabled, you can have the ReadiVoice system append (or prepend) one or more identifying strings of DTMF digits to the dialed phone number when a subscriber dials out from a conference. The switch that receives the dial-out must be configured to recognize, strip off, and properly process these digits for billing purposes.

Caution! Perform the following procedure after operational hours.

To enable dial-out billing:

1 Determine the format of the dial-out string that your switch will expect and recognize.

You can define two additional DTMF strings for each subscriber, and either or both can be outpulsed before or after the dialed number. You can define pauses, if necessary, between the dialed number and the prepended or appended digits. The total length of the string or strings to be appended/prepended is limited to 24 DTMF digits.

2 Determine how and where these identifier numbers will be stored for each subscriber.

The subscriber record provides two fields specifically for this purpose, `DialOutPreFix` and `DialOutPostFix`. You can use either or both.

3 If necessary, update your existing subscriber records with the required identifiers.

   You may want to do this using the Provisioning Stored Procedure Interface (PSPI). See the *ReadiVoice PSPI Reference*.

4 As user cnow, open the `.odprocrc` file for editing and, in the `[bridgeInt]` section, configure the `dialOutSetupString` parameter to meet your requirements (see "Dial-out Setup String Example").

5 As root, restart the ReadiVoice application (see "Stopping and Restarting ReadiVoice Processes" on page 111).

6 Make sure that future subscribers will be provisioned properly:

   – If you're using the **Dial Out Prefix** and **Dial Out Postfix** fields, make sure they're available in the Provisioning interface (see "Setting Up the Provisioning Interface" on page 60).

   – Let your ReadiVoice provisioners know how you want them to define these two fields when adding or updating a subscriber. The two fields *together* must not be more than 24 characters.

   – Make sure provisioners know that, to use this feature, subscribers must have **Dial-Out Permission** enabled.

Dial-out Setup String Example

To enable the Dial-Out Billing feature, you must add a line such as this to your `.odprocrc` file's `[bridgeInt]` section:

```
[bridgeInt]
...
   dialOutSetupString = {phoneNum},,,{dialoutPreFix},,,{dialOutPostFix}
```

The example shows the variables specific to Dial-Out Billing, but you can also use the `recorderSetupString` variables (see Table 5-7 on page 138), such as `{subId}` or `{externId}`. Each comma is a 20-millisecond pause.

The Dial-Out Billing setup string example would out-pulse three values in succession:

1 The dialed phone number, `{phoneNum}`, followed by a 60-millisecond pause.

2 The value in the subscriber's `DialOutPreFix` field, `{dialoutPreFix}`, followed by another 60-millisecond pause.

3 The value in the subscriber's `DialOutPostFix` field, `{dialOutPostFix}`.

Enabling Account Options Updating

With Account Options Updating enabled, you can use the Provisioning Stored Procedure Interface (PSPI) to retrieve records of the account options changes that your subscribers make (using the account options menu in the call flow). This lets you update your customer management database through PSPI.

At this time, the Account Options Updating mechanism only records changes made via DTMF commands; changes made through the Moderator aren't available.

This feature works as follows:

1. When a subscriber uses the account options menu (prior to conference) to change a feature setting on his or her account, the ReadiVoice system writes a record of that change to a special table, `AccOptChanges` (in addition to updating the subscriber record, of course).

   The `AccOptChanges` record identifies the subscriber, the time of the change, the subscriber field that was changed, and its old and new values.

2. Using the `GetAccOptChanges` stored procedure (see the *ReadiVoice PSPI Reference*), you retrieve the new options update records from the `AccOptChanges` table from time to time. This flags those records as processed.

3. Periodically, you must run the `cleanAccOptChanges.pl` purge script to delete the processed records from the `AccOptChanges` table.

Caution! Don't attempt to perform these tasks unless you're an experienced Informix database and UNIX system administrator. Perform the following procedure after operational hours.

To use this feature:

1. Review the `GetAccOptChanges` stored procedure description in the *ReadiVoice PSPI Reference* and ensure that you have a process in place for retrieving options update records from the `AccOptChanges` table.

2. In cnow's `crontab` file, schedule the `cleanAccOptChanges.pl` purge script to run every 24 to 48 hours. See "Purging the AccOptChanges Table" on page 110 for details of how to use this script.

3. As user cnow, open the `.odprocrc` file for editing, and find the `[bridgeInt]` section. To enable the Account Options Updating feature, set the `dtmfOptChanges` variable as follows:

   ```
   [bridgeInt]
   ...
      dtmfOptChanges = 1
   ```

4 As root, stop and restart the ReadiVoice system as described in "Stopping and Restarting ReadiVoice Processes" on page 111.

5 Use the `GetAccOptChanges` stored procedure to retrieve records from the `AccOptChanges` table periodically. To determine how frequently to call this stored procedure, balance your need for up-to-date account options data with other demands on your database and ReadiVoice system.

6 Test the system to make sure the `GetAccOptChanges` stored procedure and `cleanAccOptChanges.pl` purge script are running properly. Verify that account options updates are being:

- Written to the `AccOptChanges` table.
- Retrieved by the stored procedure.
- Purged by the purge script.

If any of these elements is *not* running properly, disable this feature immediately by repeating steps 3 and 4, but this time setting the `dtmfOptChanges` variable to zero. Then, troubleshoot your implementation of this configuration.

Caution! If you enable the Account Options Updating feature, continue to monitor the AccOptChanges table as part of your regular ReadiVoice database maintenance routine to make sure the GetAccOptChanges stored procedure and cleanAccOptChanges.pl purge script are running properly. If left unmanaged, the AccOptChanges table could slow down database operations and hinder ReadiVoice performance.

Using the Music Hold Extender Message

If many participants are waiting on hold when the subscriber arrives, there may be a short delay (a few seconds) between the time when they stop hearing the hold music and the time when they're placed into conference. You can configure the system to play a special message when the subscriber arrives if there are more than a specified number of callers on hold. The message lets participants know the system is in the process of putting them into conference.

The default prompt set includes the voice prompt file that this feature uses, `cf_music_hold_to_conf.wav`. It says, "The subscriber has joined the conference. Your conference will now begin. Please stand by." If you want different wording, you can create your own version of this file. For more information, see Appendix A.

Caution! Perform the following procedure after operational hours.

To use this "extender" message feature:

1. If you have your own version of the `cf_music_hold_to_conf.wav` file, put it into the appropriate subdirectory (`adpcm` or `g711`) of `/rahome/bridge/sound/1`, depending on format.
2. As user cnow, open the `ive.ini` file for editing, find the `[SETUP]` section, and look for an entry that reads:

   ```
   PlaySubArrivedWave = 0
   ```

   The default value of zero disables this feature.
3. Change the value after the `PlaySubArrivedWave` keyword from `0` to the number of waiting callers you want to trigger the extender message.

   If you set it to `1`., the message always gets played. If you set it to `10`, then nine or fewer callers won't trigger the message, but ten or more will.
4. Save and close the `ive.ini` file.
5. Reboot the bridges so that they read the updated `ive.ini` file and load the new `.wav` file.

   Especially for InnoVox 4000 bridges, it's important to wait until a bridge finishes booting before rebooting the next bridge.
6. To test the feature:

   a. Have enough participants dial into a conference to trigger the message (at least the number you set in `ive.ini`).

   b. Have the subscriber dial in, and verify that the participants hear the correct message.

   The bridge plays the `cf_music_hold_to_conf.wav` file to the participants immediately after identifying the subscriber.

Configuring Remote Alarm Notifications

The ReadiVoice system can notify someone when an *alarm* occurs. The ReadiVoice installation script provides the opportunity to enter one or more pager numbers, email addresses, or both, to which you want remote alarm notifications sent. This section describes how to finish setting up the pager and email notifications, test the configuration, and run the ReadiVoice Monitoring Tool that enables the notifications.

Caution! Configure and test alarm notifications only after operational hours.

Pager Notification Setup

All ReadiVoice systems include one modem (and cable) to permit dial-up remote maintenance. A second modem, dedicated to the remote alarm function, is optional.

However, remote alarm configuration doesn't *require* a dedicated modem for pager notification. You can configure remote alarm notification to use the same modem as the dial-up remote maintenance function. Keep in mind, however, that alarm notifications to pagers will fail if that modem is in use at the time.

By default, the remote alarm function is configured in the system files to use the dedicated remote alarm modem connected to serial port B. Install this modem using the cable provided or your own full 25-pin serial cable.

The `/etc/remote` file specifies the serial port to use and the modem-specific settings. If you need to change any of the default settings, see the `remote(4)` man page for complete details on the configuration settings in the `/etc/remote` file.

Figure 5-2 shows the default configuration lines in the `/etc/remote` file for a system using the Polycom-supplied dedicated modem.

Figure 5-2 *The default alarm modem settings in the /etc/remote file*

```
pager:\
:pn=@:tc=cuaa:
cuaa:\
:el=^D^U^C^S^Q^O@:du:at=hayes:ie=#$%:oe=^D:br#38400
:tc=dialers:
```

The `/etc/phones` file specifies the number or numbers to call for pager alerts. When you enter pager numbers during the installation, the install script creates an entry in this file for each. You can edit this file manually to finish configuring the entries, make changes, or if you skipped alarm configuration during the installation.

Each line in the `/etc/phones` file takes the form

```
pager PHONENUMBER,,,,,,_ALARMCODE_
```

where *PHONENUMBER* is the complete phone number needed to reach the pager, including any digits required to access an outside line. If the pager service requires a PIN code, include it as well.

The *PHONENUMBER* string can contain valid DTMF keys (the digits 0 through 9, #, and *) and commas.

Commas represent modem-specific delays, typically one second each. If the modem must dial an 8 or 9 to access an outside line, you may need to insert a comma or two after that number to allow enough time for accessing the line. Also, different paging service providers may require more or less delay to complete a call or process a PIN code. Therefore, you must adjust the number of commas properly for each pager entry. This may be a trial-and-error process. See "Testing Remote Alarm Notification" on page 133.

When an alarm occurs, the placeholder `_ALARMCODE_` is replaced with the actual numeric alarm code, which the modem sends as DTMF digits. *Don't* change the keyword `pager` or the placeholder `_ALARMCODE_`. Modify only the dialing string between these two items.

Here are two sample pager entries (the second includes a PIN code):

```
# Page George in New York office
pager 9,,18005551212,,,,,_ALARMCODE_

# Page local support through on call pager number.
pager 8,6466672,,,1455#,,,,,,,_ALARMCODE_#
```

Some paging services ask callers to press the pound key after entering the number to be sent. As the second example shows, you can accommodate this by putting a pound sign after `_ALARMCODE_`.

Email Notification Setup

You can have the ReadiVoice system send alarm notifications to designated email addresses, either instead of or in addition to telephone numbers.

Assuming that your Sun is set up properly for UNIX mail, you can send mail to any known user (such as `root`) on your network. To send an email elsewhere (such as an Internet email address), your network must have an SMTP server with Internet access. Contact your network specialist about configuring your network for email notification.

The `/rahome/bin/email` file specifies the addresses to which to send email alerts. When you enter email addresses during the installation, the install script creates an entry in this file for each. You can edit this file manually to make changes or if you skipped alarm configuration during the installation. Each line in the file contains a single email address. For instance, it might contain the following:

```
root
cnow
george_w@confservices.com
```

Testing Remote Alarm Notification

To verify that remote alarm notification is working properly and to adjust the timing in the pager notifications, you need to generate alarm notifications for testing. You can do so at the command prompt. To create a test alarm, sending the alarm code `123456789`, enter this command:

`/rahome/bin/alarm.sh 123456789`

The pagers you entered should receive the page. Depending on the paging service, this may take a few minutes.

To test the entire process, including alarm code generation and both email and pager notification, create loss of capacity events. The simplest approach is to remove spans manually until you reach the capacity threshold for the bridge or system. For instance, with the default threshold setting, you can generate a bridge capacity alarm for a 480-port T1 bridge by removing four or more spans.

If you want to generate a system capacity alarm for a multiple-bridge system, you may want to temporarily increase the threshold setting first.

The default settings for alarm monitoring work well for most situations, but you can modify them if necessary. The sections that follow describe how to change the alarm settings, run the ReadiVoice Monitoring Tool, and interpret the alarm codes.

The ReadiVoice Monitoring Tool must run continuously in order to provide alarm monitoring and remote alarm notification.

Changing the Alarm Settings

To change the default alarm settings:

1 Establish a Telnet connection to the Sun server and log in as cnow.

2 Change directory to `/rahome/bin`.

3 Open the `.odprocrc` file for editing and locate the alarm section (Figure 5-3).

4 Referring to Table 5-4, change the values that you want to change, being careful not to change anything else in the file.

5 Save and close the `.odprocrc` file.

Figure 5-3 *The alarm settings in the .odprocrc file*

```
[alarm]

   system_id  =  01
   check_mibs  =  0
   interval  =  900
   sysmax_capacity  =  480
   systreshold  =  80
   brgmax_capacity_1  =  480
   brgtreshold_1  =  80
   brgmax_capacity_2  =  240
   brgtreshold_2  =  80

[]                          // [alarm]
```

Table 5-4 *Alarm section parameters of `.odprocrc` file*

Keyword	Description
`system_id`	Uniquely identifies your ReadiVoice system. *Don't* change this number.
`check_mibs`	Boolean. If this is set to 1, at each polling interval, the `ra_monitor.pl` script checks the MIB variables listed in the `ra_monitor.mibs` file and prints them to standard output (the terminal window in which it's running).
`interval`	Interval in seconds between polling for alarms. The default value of 900 (15 minutes) works well for large or busy systems. For a single-bridge or moderately busy system, you can reduce this to 500 – 700 seconds. For short-term testing or troubleshooting purposes, you can set this value to less than 20 seconds; but, you may notice that the ReadiVoice system slows down at this setting.
`sysmax_capacity`	Maximum capacity (number of ports) of your system. *Don't* change this number.
`brgmax_capacity`	Maximum capacity (number of ports) of an individual bridge. *Don't* change this number.

Table 5-4 *Alarm section parameters of `.odprocrc` file (continued)*

Keyword	Description
`systreshold`	Percentage of maximum system capacity at which a capacity alarm is triggered. The default value is 80, which means an alarm occurs if the number of available ports in your system drops to 80% or less of sysmax_capacity. Change this number to the threshold you want to use for system capacity alarms.
`brgtreshold`	Percentage of maximum bridge capacity at which a capacity alarm is triggered. The default value is 80, which means an alarm occurs if the number of available ports in an individual bridge drops to 80% or less of brgmax_capacity. Change this number to the threshold you want to use for bridge capacity alarms.

Continuously Monitoring for Alarms

To monitor your system continuously for alarms:

1. On the Sun server, log in as root, open an XTerm window, and switch user to cnow. You can't use a Telnet session for this purpose.
2. Change directories to `/rahome/bin`.
3. Type **`./ra_monitor.pl`** and press ENTER.

   The ReadiVoice Monitoring Tool starts running. See Figure 5-4.

Leave this window open as long as you want to monitor the system. If you close this XTerm session, monitoring stops.

Alternatively, you can run the monitoring tool as a background process by entering **`nohup ./ra_monitor.pl &`**. Then you can close the XTerm window.

Figure 5-4 *ReadiVoice Monitoring Tool running*

```
$
              \///
              (o o)
-----------ooO-(_)-Ooo-----------------------

  Welcome to Monitoring Tool


  Mon Mar 29 14:34:53 MST 1999

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

ebrgStatus.0 = bridgeUp(1)

brgStatus.1 = bridgeUp(1)
$ █
```

Interpreting the Alarm Codes

When an alarm condition occurs, the ReadiVoice Monitoring Tool sends a numeric alarm code to designated pager numbers and email addresses. The numeric code consists of from six to ten digits that identify the source of the alarm and its nature. The tables below help you understand the alarm codes.

Table 5-5 describes the format for the alarm codes.

Table 5-5 *Alarm code format*

First 2 digits	System ID (customer specific).
Next 2 digits	Error code.
Next 2 digits	Number of bridges logged in.
Next 2 digits	Bridge with problem; only used with error code 06.
Next 2 digits	Bridge capacity threshold; only used with error code 06.

Table 5-6 shows the possible error code values (the third and fourth digits in the alarm code).

Table 5-6 *Error code descriptions*

Code	Description
01	CACS is down.
02	Mplex server is down (can't run Operator interface).
03	SNMP daemon is down (no monitoring because system status isn't known).
04	Non-busyout bridge logged out (bridge crash).
05	System capacity has dropped below threshold (by default, 80% of maximum).
06	Bridge capacity has dropped below threshold (by default, 80% of maximum). This indicates spans in alarm state.

For example, if you receive an alarm code of 0806020180, it tells you that on your system (08), there is a bridge capacity alarm (06). Two bridges are logged in (02) and Bridge 1 is the one in alarm state (01). Its capacity has dropped below the threshold of 80% (80).

Enabling Conference Recording

You can enable your subscribers to record their conferences by configuring your ReadiVoice system to connect to a recording device or service using a dial-out connection.

Recording can be initiated using a designated touch-tone command or a button on the Moderator interface. The sections that follow describe how to configure and enable a recording connection and how it works.

Caution! Configure and test conference recording only after operational hours.

Enabling Conference Recording Connections

To set up your system for conference recording connections:

1. Verify that the voice prompt files used for recording are appropriate for your business needs or update them as necessary. The voice prompt files are listed in "Voice Prompts for Conference Recording" on page 138.

   Polycom recommends professional recording of all voice prompt files. For file format requirements, see Appendix A.

2. Determine the telephone number that the ReadiVoice system can use to connect to the external recording device or service and the DTMF key sequence that the ReadiVoice system can use to initiate recording.

3. As user cnow, edit the `.odprocrc` file to set up the `recorderPhone` and `recorderSetupString` parameters to implement the recording configuration determined in 2. See "Recording Configuration Variables in the .odprocrc File" on page 138.

4. Save your changes to `.odprocrc`. Then, as root, stop and restart the ReadiVoice system (see "Stopping and Restarting ReadiVoice Processes" on page 111) to load the new settings.

5. Test the recording capability:

   a. Set up or open a test subscriber account.

   b. In Provisioning, turn on **Recorder Dial Out** for the test account.

   c. If you used the `{externId}` variable in the setup string, make sure the **Subscriber External ID** is populated.

   d. Save the changes to the test account.

   e. Start a conference as the test subscriber.

   f. Use the touchtone command for your system to start recording.

   g. Speak into the conference to create a recording that you can verify.

   h. End the conference.

6 Verify that the test conference was recorded by listening to the recording. If the test conference was recorded, continue to the next step. If not, review the previous steps or call your Polycom Global Services representative for assistance.

7 Instruct provisioners to turn on **Recorder Dial Out** for subscribers who need to be able to record conferences.

Voice Prompts for Conference Recording

The voice prompt files used during the recording process are:

```
rec_change_failed.wav
rec_conf_full.wav
rec_not_enabled.wav
rec_part_join_reminder
rec_rejoin_reminder.wav
rec_setup_failed.wav
rec_start_prompt.wav
rec_started.wav
rec_stop_prompt.wav
rec_stopped.wav
rec_subs_join_reminder
rec_wait.wav
```

For the default prompt text and file format requirements, see Appendix A. Figure A-9 on page 213 shows the call flow.

Recording Configuration Variables in the .odprocrc File

To enable conference recording, the `recorderPhone` and `recorderSetupString` values in the `[modules] [bridgeInt]` section of `.odprocrc` must be specified:

- For `recorderPhone`, specify the telephone number of the recording device or service.
- For `recorderSetupString`, use valid DTMF keys (**0-9**, *****, and **#**) and the variables listed in Table 5-7 to define a setup string that will uniquely identify each recording of each conference.

See the example shown in Figure 5-5 on page 140 for one implementation.

Table 5-7 *Variables you can use in recorderSetupString*

Variable	Description
`{subId}`	Subscriber ID.
`{externId}`	Contents of External ID field from SUBSCRIBERINFO table.
`{ccnum}`	Credit card number from BILLINGINFO table.

Table 5-7 *Variables you can use in recorderSetupString*

Variable	Description
`{ccmonth}`	Credit card expiration month.
`{ccyear}`	Credit card expiration year.
`{confStart}`	Conference start time in UNIX seconds.
`{now}`	Time in UNIX seconds when the recording device or service is dialed.
`{accesscode}`	Subscriber's access code from ASSIGNEDPHONES table.
`{securitycode}`	Conference security code set by the subscriber for the conference, if any.
`{accessnum}`	Access phone number that the subscriber used to dial into the system (empty if the system dialed out to the subscriber).
`{partpin}`	Participant password from SUBSCRIBERINFO table (if subscriber has two-password call flow; otherwise, field is empty).
`{subpin}`	Subscriber password from SUBSCRIBERINFO table.
`{subgroupId}`	Subscriber group ID from SUBSCRIBERINFO table.
`{subscribernote1}`	Contents of first optional subscriber note field (Subscriber User Field A in Provisioning) from SUBSCRIBERDETAIL table.
`{subscribernote2}`	Contents of second optional subscriber note field (Subscriber User Field B in Provisioning) from SUBSCRIBERDETAIL table.
`{billingnote1}`	Contents of first optional billing note field (Billing User Field A in Provisioning) from the BILLINGINFO table.
`{billingnote2}`	Contents of second optional billing note field (Billing User Field B in Provisioning) from BILLINGINFO table.
`{confId}`	Conference ID (from CDR_POST_CONF table) of conference from which the recorder dial-out is being initiated.

Since the `recorderSetupString` is outpulsed as DTMF digits, data referenced by the variables that you use must be numeric. If a retrieved variable string contains a non-numeric character, the system omits the entire string for that variable from the outpulsed data.

In particular, be aware that the ReadiVoice system performs no validation on the two subscriber note fields or the two billing note fields during the provisioning process. In most system configurations, it also doesn't validate the External ID field. If you intend to use any of these fields for recorder setup information, it's up to you to ensure that they contain numeric data and that your provisioning process populates the fields properly.

How Conference Recording Works

Figure 5-5 shows an example of `.odprocrc` configuration settings for conference recording.

Figure 5-5 *Example of conference recording settings in the `.odprocrc` file*

```
[modules]
   [bridgeInt]
     ...
     recorderPhone = 5551212
     recorderSetupString =1,,4,,5*{accesscode}#*{confStart}##
     ...
   []
[]
```

In a system using the example settings shown, the following sequence occurs when a subscriber selects the recording option:

1. The subscriber selects the option to record the conference in one of two ways:
   - Entering the touchtone command (default is ***2**).
   - Clicking the **Record** button in the Moderator.
2. The ReadiVoice system dials the phone number specified by `recorderPhone`. In the example, this is 5551212.
3. The system sends the DTMF tones specified by `recorderSetupString`, substituting actual DTMF tones for the variables. With the example configuration shown in Figure 5-5, the system sends (with DTMF digits replacing the variables described in brackets):

   1 (pause) **4** (pause) **5 *** [subscriber's access code] **# ***
   [start time of conference] **# #**

   In this example, the system uses the DTMF sequence **1,,4,,5** to maneuver through an IVR menu, with pauses between DTMF keys to allow the IVR platform to process them (each comma represents a 20-millisecond pause). It sends *** and **#** to delimit the fields and **##** to mark the end of the setup string to the recording device. This is one possible implementation.
4. Conference recording begins.
5. Recording ends when the conference ends or when the recorder telephone line is disconnected.

Enabling SNMP Logging

The SNMP logging function writes system usage and performance data to log files on an ongoing basis. For more information about SNMP, logging, and the log files, see .

Caution! Perform the following procedure only after operational hours.

To turn on SNMP logging:

1. As cnow, open `.odprocrc` for editing.
2. In the `[snmpMgr]` section of the file, edit the logging parameters as described in Table 5-8. Figure 5-6 shows a sample `[snmpMgr]` section.
3. Save your changes to `.odprocrc`. Then, as root, stop and restart the ReadiVoice system (see “Stopping and Restarting ReadiVoice Processes” on page 111) to load the new settings.

Table 5-8 *SNMP logging parameters of the `.odprocrc` file*

Parameter	Value/Description
`logPrint`	1 Enables SNMP logging. 0 Disables SNMP logging.
`logInterval`	*n* Polling interval, in seconds, at which log file records are written (minimum interval is 600 seconds).
`snmplogfile`	*`filename`*`.log` The full path and name of the SNMP log file (default is `/rahome/bin/snmp/snmp.log`).
`histogramlogfile`	*`filename`*`.log` The full path and name of the histogram log file (default is `/rahome/bin/snmp/histogram.log`).

Figure 5-6 *The SNMP logging settings in `.odprocrc`*

```
[machines]
...
   [snmpMgr]
     ...
     logPrint = 1
     loginterval = 600
     snmplogfile = /rahome/bin/snmp/snmp.log
     histogramlogfile = /rahome/bin/snmp/histogram.log
     ...
   []
...
[]
```

Changing the Web Server Port

This procedure allows you to change the Web server port used by the ReadiVoice HTML pages (Administration, Provisioning, Operator login, and Moderator) from port 80 to something else.

This procedure doesn't modify the port used by Java applets, such as the Operator, Channel State Monitor, and SNMP Monitor.

By default, the script moves the Web server to port 40004, but you can specify any port.

Caution! Perform the following procedure only after operational hours.

To change the Web server port:

1 Log into the ReadiVoice CACS as root and change to the `/rahome/utils` directory.

2 Make sure the `webport` script has 755 file permissions.

3 Type **`./webport`** and press ENTER to run the script.

4 When prompted for what port to use, do one of the following:

   – Press ENTER to accept the default, `40004`.

   – Type a different port number and press ENTER.

5 When prompted for the host name, do one of the following:

   – Press ENTER to accept the host name shown in brackets.

   – Type a different host name and press ENTER.

6 Watch the messages that scroll by as the script stops the Web server, changes the port, and restarts the Web server on the new port.

   If the script succeeds, it displays a final message like this:

   ```
   Startup: listening to http://host, port portnum as web
   ```

   In place of *`host`* and *`portnum`*, you'll see the host name and port number you specified.

   If there is an error binding to the port you specified, the script displays a message like this:

   ```
   Unable to bind to port portnum - error 125
   ```

   This means that the port is already in use by another process.

7 If the port you specified is already in use, check the `/etc/services` file to see which ports are in use, choose another one, and rerun the script. If you need help choosing a port or if you see other error messages, call your Polycom Global Services representative.

8 If the script succeeded, test the URL that includes the new Web server port number by pointing a browser to:

`http://`*`hostip`*`:`*`portnum`*`/index2.html`

Use the IP address of the SUN (CACS) for *`hostip`* and the port number you specified for *`portnum`*. For example, an actual URL might be:

`http://127.0.0.1:40004/index2.html`

9 Ensure that users of this system access the new URL.

Enabling Operators to Unlock Conferences

If your operators use the ReadiVoice Operator application for Windows (which we refer to simply as the Windows Operator), they can lock conferences. By default, however, the Windows Operator is configured so that they can't unlock conferences.

If you want your operators to be able to unlock conferences, you can easily enable this feature. Note that you can enable operator unlock for a specific operator station (PC), but not for a specific operator or operators. To enable this feature globally, you must make the change on each operator station.

To enable operator unlock on an operator station:

1 Log into the operator station as a user with administrator rights and make sure the Windows Operator isn't running.

2 In Windows Explorer (or another file manager), navigate to the Windows Operator installation folder. Typically, this is:

`C:\Program Files\Polycom ReadiVoice\WinOp.Installer\`

3 Find the `Operator.exe.config` file and open it in Notepad or another text editor.

This is an XML file that defines a number of key-value pairs. The one you need to change looks like this:

```
<add key="WinOp.Client.AllowUnlock" value="false" />
```

4 Change the valueof the key to true, so that the line reads:

```
<add key="WinOp.Client.AllowUnlock" value="true" />
```

5 Save and close the file. Log out of the operator workstation.

The next time an operator logs in and starts the Windows Operator, it reads the modified configuration file and enables conference unlock.

7

Customizing & Branding Your ReadiVoice System

This chapter describes some of the customizing and branding options in the ReadiVoice system. It assumes that you know Solaris UNIX, Sun servers, personal computers, software configuration, telephony, and networking.

Customizing Touchtone Commands

The touchtone (DTMF, or dual tone multi-frequency) commands that the ReadiVoice system recognizes are defined in the bridge initialization file for the system, `/rahome/bridge/scripts/ive.ini`.

Before changing your system's touchtone commands, read:

- "Rules and Guidelines for Customizing Commands"
- "Procedure for Customizing Commands" on page 156
- "Detailed Information for Specific Touchtone Features" on page 158

Rules and Guidelines for Customizing Commands

Figure 7-1 shows the default `ive.ini` file's `[DTMF_CMDS]` section. For each entry, the touchtone key sequence is to the left of the equal sign, and the command to which it's assigned is to the right. Entries preceded by two slashes (`//`) are commented out (disabled). You can enable or disable commands, or change the touchtone key sequences that invoke the commands.

Any changes you make to the entries in the `[DTMF_CMDS]` section of the `ive.ini` file *absolutely must* obey the following rules:

- Change *only* the left side of the equal sign, which specifies the DTMF key sequence assigned to a command. The command names, beginning with `eCMD_`, must not be modified in any way.
- Each key sequence must be unique (not assigned to more than one command). If the same key sequence is assigned to two commands, one of them must be commented out (disabled).
- Each key sequence must consist of from one to three DTMF keys.
- Only characters that appear on standard telephone keypads are valid DTMF keys. They are **0**, **1**, **2**, **3**, **4**, **5**, **6**, **7**, **8**, **9**, *****, and **#**.

When you make DTMF command changes, you also need to change the affected voice prompts. For instance, any change in commands will require at least a new "help" prompt, which describes the available commands. See Appendix A for detailed information about the voice prompts used in the ReadiVoice system.

Procedure for Customizing Commands

To change the DTMF key sequences associated with commands or to enable or disable commands:

1. Check to determine which voice prompt (`.wav`) files are impacted by the command change you want to make. Replace the `.wav` files if necessary.

   We recommend that `.wav` files be recorded professionally. Contact your Polycom Global Services representative if you want help in obtaining appropriate `.wav` file recordings.

2. Log into the CACS as root and switch user to cnow.
3. Open the `ive.ini` file for editing and find the `[DTMF_CMDS]` section near the end of the file.

Figure 7-1 *`[DTMF_CMDS]` section of default `ive.ini` file*

```
//////////////////////////////////////////////////////////////////////
//  The following entries configure the DTMF commands for users in
//  conference.
// NOTE: All DTMF commands must be unique, duplicate values will cause
// ive.ini read error
//////////////////////////////////////////////////////////////////////
[DTMF_CMDS]
   ** = eCMD_HELP
   *0 = eCMD_OP_REQUEST
   00 = eCMD_CONF_REQUEST
   *1 = eCMD_DIAL_OUT
   *2 = eCMD_CONF_RECORD
   // *2 = eCMD_ROLL_CALL_CONF
   *3 = eCMD_CONF_CONFIG
   // *4 = eCMD_LOCK_TOGGLE
   *4 = eCMD_CONF_LOCK
   *5 = eCMD_CONF_UNLOCK
   // *6 = eCMD_MUTE_TOGGLE
   *6 = eCMD_MUTE
   *7 = eCMD_UNMUTE
   *8 = eCMD_CONF_CONTINUE
   *9 = eCMD_ROLL_CALL_PRIV
   *# = eCMD_PART_COUNT
   ## = eCMD_MUTE_ALL
   // ## = eCMD_MUTE_ALL_TOGGLE
   99 = eCMD_UNMUTE_ALL
   // #1 = eCMD_LISTEN_ONLY_TOGGLE
   #1 = eCMD_LISTEN_ONLY
   #2 = eCMD_UNLISTEN_ONLY
   #5 = eCMD_WR_PROCESS
   #6 = eCMD_WR_ANNOUNCE_TOGGLE
   #8 = eCMD_CONF_END_CONF
   1  = eCMD_WR_PART_TO_CONF
   2  = eCMD_WR_PART_IGNORE
   3  = eCMD_WR_PART_DISCONNECT

[]
```

4. To disable a command, comment out its line by putting two slashes (//) at the beginning of the line.

   Don't delete lines that you don't want to use. Comment them out so that they're ignored, but you don't lose the syntax information. This allows you to enable them easily later by just removing the slashes.

5 To enable a disabled command, remove the two slashes (//) at the beginning of the line.

Caution! Some commands are mutually exclusive. They can be enabled only if the alternatives are disabled. See "Detailed Information for Specific Touchtone Features" for discussions of these commands.

6 To change the DTMF key sequence that invokes a command, edit the two characters to the left of the equal sign for that command.

7 When you're finished making changes, save and close the `ive.ini` file.

8 Reboot the bridges so that they read the updated `ive.ini` file and load the new `.wav` files.

9 Set up a test conference and verify that the touchtone commands you modified work as intended and that the voice prompts are correct.

10 Make sure your teleconferencing users are aware of the changes.

Detailed Information for Specific Touchtone Features

This section provides more detailed information about the following touchtone command features and how they can be implemented:

- Lock/Unlock
- Mute/Unmute
- Mute All / Unmute All
- Listen Only / Unlisten Only
- Private and Conference Roll Call

Lock/Unlock

Conference locking and unlocking can be controlled either with two separate commands or with a single command that toggles the lock state of the conference.

The default `ive.ini` file (Figure 7-1 on page 157) enables the two-command option; **`*4`** locks a conference and **`*5`** unlocks it. The relevant lines look like this:

```
//*4 = eCMD_LOCK_TOGGLE
*4 = eCMD_CONF_LOCK
*5 = eCMD_CONF_UNLOCK
```

With the standard toggle option, pressing **`*4`** locks a conference and pressing **`*4`** again unlocks it. To enable this configuration, change the relevant lines to look like this:

```
*4 = eCMD_LOCK_TOGGLE
//*4 = eCMD_CONF_LOCK
//*5 = eCMD_CONF_UNLOCK
```

Caution! If you enable the eCMD_LOCK_TOGGLE command, you must disable both the eCMD_CONF_LOCK and eCMD_CONF_UNLOCK commands.

You can, of course, change the DTMF key sequence(s) used in either configuration, but you'll also have to change the relevant voice prompts (see).

Mute/Unmute

Muting and unmuting of individual lines can be controlled either with two separate commands or with a single command that toggles the mute state of the channel.

The default `ive.ini` file (Figure 7-1 on page 157) enables the two-command option; **`*6`** mutes the line of the person entering the command and **`*7`** unmutes it. The relevant lines in `ive.ini` look like this:

```
//*6 = eCMD_MUTE_TOGGLE
*6 = eCMD_MUTE
*7 = eCMD_UNMUTE
```

With the standard toggle option, pressing **`*6`** mutes the line of the person entering the command and pressing **`*6`** again unmutes it. To enable this configuration, change the relevant lines in `ive.ini` to look like this:

```
*6 = eCMD_MUTE_TOGGLE
//*6 = eCMD_MUTE
//*7 = eCMD_UNMUTE
```

Caution! If you enable the eCMD_MUTE_TOGGLE command, you must disable both the eCMD_MUTE and eCMD_UNMUTE commands.

You can, of course, change the DTMF key sequence(s) used in either configuration, but you'll also have to change the relevant voice prompts (see).

You can also customize the system's mute confirmation behavior. See "Customizing Confirmation Sounds" on page 162.

Mute All / Unmute All

Muting and unmuting of all participants can be controlled either with two separate commands or with a single command that toggles the mute state of the conference.

The default `ive.ini` file (Figure 7-1 on page 157) enables the two-command option; **##** mutes all the ordinary participants (not subscriber or operator) in the conference and **99** unmutes them. The relevant lines in `ive.ini` look like this:

```
## = eCMD_MUTE_ALL
// ## = eCMD_MUTE_ALL_TOGGLE
99 = eCMD_UNMUTE_ALL
```

With the standard toggle option, pressing **##** mutes all the ordinary participants and pressing **##** again unmutes them. To enable this configuration, change the relevant lines in `ive.ini` to look like this:

```
// ## = eCMD_MUTE_ALL
## = eCMD_MUTE_ALL_TOGGLE
// 99 = eCMD_UNMUTE_ALL
```

Caution! If you enable the eCMD_MUTE_ALL_TOGGLE command, you must disable both the eCMD_MUTE_ALL and eCMD_UNMUTE_ALL commands.

You can, of course, change the DTMF key sequence(s) used in either configuration, but you'll also have to change the relevant voice prompts (see).

You can also customize the system's mute confirmation behavior. See "Customizing Confirmation Sounds" on page 162.

Listen Only / Unlisten Only

Turning listen only mode on and off can be controlled either with two separate commands or with a single command that toggles the listen-only state of the conference.

The default `ive.ini` file (Figure 7-1 on page 157) enables the two-command option; **#1** puts the conference into listen only mode (all ordinary participants are muted and can't unmute themselves) and **#2** turns off listen only mode (enabling ordinary participants to speak again). The relevant lines in `ive.ini` look like this:

```
// #1 = eCMD_LISTEN_ONLY_TOGGLE
#1 = eCMD_LISTEN_ONLY
#2 = eCMD_UNLISTEN_ONLY
```

With the standard toggle option, pressing **#1** turns on listen only mode and pressing **#1** again turns it off. To enable this configuration, change the relevant lines in `ive.ini` to look like this:

```
#1 = eCMD_LISTEN_ONLY_TOGGLE
// #1 = eCMD_LISTEN_ONLY
// #2 = eCMD_UNLISTEN_ONLY
```

Caution! If you enable the eCMD_LISTEN_ONLY_TOGGLE command, you must disable both the eCMD_LISTEN_ONLY and eCMD_UNLISTEN_ONLY commands.

You can, of course, change the DTMF key sequence(s) used in either configuration, but you'll also have to change the relevant voice prompts (see).

Private and Conference Roll Call

The ReadiVoice system supports two different implementations of the roll call feature, which plays the recorded names of participants, along with a participant count:

- *Private roll call* plays the roll call only to the requesting line. It's available to both the subscriber and participants.
- *Conference roll call* plays the roll call to the entire conference. It's available to only the subscriber.

The default `ive.ini` file (Figure 7-1 on page 157) enables only the private roll call option, leaving the conference roll call command commented out (disabled). The relevant lines in `ive.ini` look like this:

```
// *2 = eCMD_ROLL_CALL_CONF
...
*9 = eCMD_ROLL_CALL_PRIV
```

You can have either or both of these commands enabled. Of course, no two commands can use the same DTMF key sequence, so if you want to enable the conference roll call option you must do one of the following:

- Disable the private roll call command and assign its key sequence (***9**) to the conference roll call command.
- Disable the conference record command so that the private roll call command can use its key sequence (***2**).
- Assign some other unused key sequence (for instance, **#9**) to the conference roll call command.

You'll also have to change the relevant voice prompts (see Appendix A).

Customizing Confirmation Sounds

The `[CustomSounds]` section in `ive.ini` controls how the system confirms the following changes:

- Mute and unmute of an individual line.
- Mute all and unmute all.
- Listen only on and off.
- Waiting room announcements/reminders on and off.

Each of these changes can be set to a tone confirmation (two-tone sequence), a specific confirmation `.wav` file, or nothing at all. Confirmations of mute and unmute are played to the individual line. Confirmations of the other changes are played to the conference.

The default settings provide tone confirmations for mute and unmute and for the toggling of waiting room announcements/reminders, but don't provide any confirmations for the global mute and listen only commands. The relevant lines in `ive.ini` look like this:

```
[CustomSounds]
   [Mute]
      MUTE_SOUND = TONE
      UNMUTE_SOUND = TONE
      MUTE_ALL_SOUND = NONE
      UNMUTE_ALL_SOUND = NONE
      LISTEN_ONLY_SOUND = NONE
      UNLISTEN_ONLY_SOUND = NONE
   []
   [WaitingRoom]
      CONF_ANNOUNCE_ON = TONE
      CONF_ANNOUNCE_OFF = TONE
   []
[]
```

To change any of these settings, change the value after the equal sign. The following values are valid:

`TONE` to play a two-tone DTMF sequence.

`NONE` to play no sound at all (no confirmation).

`filename``.wav` to play the voice prompt file specified.

If you specify a `.wav` file to play, the ReadiVoice system looks for that file first in the directory containing the subscriber's prompt set (for instance, this is `/rahome/bridge/sound/2/adpcm` for Prompt Set 1 on a system with InnoVox 480 bridges). If the file isn't there, it looks in the directory that contains the default prompt set installed with the ReadiVoice system (`/rahome/bridge/sound/1/adpcm` on a system with InnoVox 480 bridges).

The default prompt set installed with the ReadiVoice system contains six `.wav` files with spoken messages designed to work with these commands:

```
global_muted.wav
global_unmuted.wav
private_muted.wav
private_unmuted.wav
wr_notification_off.wav
wr_notification_on.wav
```

For more information about these files, their default contents, and how to customize them, see Appendix A.

Customizing the Moderator Interface

By default, the ReadiVoice Moderator displays a Polycom logo image file on both the login/logout page and the Moderator interface page. You can customize the Moderator in two ways:

- Replace the Polycom logo image file to brand both the login/logout page and the Moderator interface page with your company logo.
- Install additional image files to customize the Moderator interface page for specific subscriber groups.

Branding the Moderator with Your Company Logo

The default image file is the image file used for everyone except the subscriber groups that you customize.

The default image file is:

```
/rahome/ns-home/docs/HTMLMOD/default/images/moderator/corporate_logo.png
```

Its size is 35x146 pixels (HxW).

In order to install a new image file, you must be logged into the CACS server as root.

To replace the existing (Polycom logo) image file with your own company's default image file:

1 Create the replacement image file you want to use as the default file.

This procedure is simplest if the replacement image is a `.png` file of the same or similar size. But, you can use a `.gif` or `.jpg` file. Other file formats are also possible (such as `.swf`), but may impose browser limitations or require plug-ins. If your replacement differs in size from the one we provide, be sure to test it to ensure that the appearance is acceptable.

2. Use file transfer protocol (`ftp`) to copy the replacement image file to a local directory on the CACS server. Be sure to use binary transfer mode.
3. Open a Telnet session to the CACS server, log in as cnow, and switch user to root.
4. Make a backup of the existing default image file in case you need to restore it.
5. If you're replacing the existing image file with another `.png` file, rename your replacement file `corporate_logo.png` and copy it to the `/rahome/ns-home/docs/HTMLMOD/default/images/moderator/` directory, overwriting the existing file.

   This is by far the easiest solution, and it doesn't require any special precautions when your ReadiVoice system is upgraded.
6. If your replacement file is a different file format, or if you don't want to use the same file name for some other reason, do the following:
   a. Copy the replacement file to a location in `/rahome/ns-home/docs/` (`/rahome/ns-home/docs/HTMLMOD/default/images/moderator/` is OK).
   b. Change directories to `/rahome/ns-home/docs/HTMLMOD/default/`.
   c. Open `moderator.html` for editing and find the string `corporate_logo.png` in two different places (in both places, it's the `src` parameter in an `<IMG>` tag).
   d. Change both occurrences of `corporate_logo.png` to the name of the replacement file. Change the path to the replacement file if necessary.

      The web server's document root directory is `/rahome/ns-home/docs/`, so the path to the replacement file must be relative to that directory.
   e. Save and close `moderator.html`.

   If you customize the Moderator by editing the `moderator.html` file, remember that the HTML file is likely to be replaced whenever you upgrade your ReadiVoice system. Therefore, you may have to repeat this process each time your system is upgraded.
7. In your browser, access the Moderator to verify that the new image file appears.

Customizing the Moderator for a Specific Subscriber Group

You can further customize the Moderator interface for a specific subscriber group by installing a group-specific image file. On the Moderator interface page (but not the login page), the group-specific file replaces the default image file for members of that group.

To customize the Moderator for a specific subscriber group:

1. In the System Administration Subscriber Groups page, determine the Group ID of the subscriber group.
2. Create the image file you want to use for this subscriber group. Give it a clearly identifiable name, such as `group#.png`, where *#* is the Group ID of the subscriber group you want to customize.

   We recommend using a `.png` file of the same size as the default, but you can use other formats, such as `.gif` or `.jpg`. If your replacement differs in size from the default, be sure to test it to ensure that the appearance is acceptable.
3. Use file transfer protocol (`ftp`) to copy the replacement image file to a local directory on the CACS server. Be sure to use binary transfer mode.
4. Open a Telnet session to the CACS server, log in as cnow, and switch user to root.
5. If it doesn't already exist, create the directory `/rahome/ns-home/docs/moderator2/group_custom`. Make sure the directory is owned by `root:web` and has permissions set to `750`.
6. Change to the `/rahome/ns-home/docs/moderator2/images` directory and create a new subdirectory named `group#`, where *#* is the Group ID of the subscriber group you want to customize. Make sure the directory is owned by `root:web` and has permissions set to `750`.
7. Copy the custom image file for the group to the new `group#` subdirectory. Make sure the file is owned by `root:web` and has permissions set to `640`.
8. From the `/rahome/ns-home/docs/moderator2` directory, copy the `default_group.ini` file to `group_custom/group#.ini`, where *#* is the Group ID of the subscriber group you want to customize. Make sure the file is owned by `root:web` and has permissions set to `640`.
9. Change to the `group_custom` directory and open the new `group#.ini` file for editing.
10. Replace the reference to `images/workspaceimage.jpg` with `images/group#/group#.png` (where `group#` is the subdirectory you created in 6 and `group#.png` is the name of the customized image). Then, save and close the file.
11. In your browser, log into the Moderator as a subscriber from the group you customized and verify that the new image appears.

Repeat the above procedure to customize another subscriber group.

Working with Prompt Sets

Prompt sets are sets of `.wav` audio files (in the WAVE format; see "Details About the WAVE File Format" on page 175) that the ReadiVoice system uses for the messages and prompts played to subscribers and participants. When you create a subscriber group, you select a prompt set for the group. The ReadiVoice system supports the use of up to 20 prompt sets. This permits you, for example, to provide customized foreign language prompts for certain customers.

Custom *welcome messages* (the initial greetings that callers hear) are handled differently from the prompt set messages. See "Customizing Greetings and Related Messages" on page 169 for more information.

The installation process puts the default voice prompt files into the following locations:

- `/rahome/bridge/sound/1/adpcm` contains the `.wav` files in ADPCM format (for InnoVox 480 bridges).
- `/rahome/bridge/sound/1/g711` contains the `.wav` files in µ-law format (for InnoVox 4000 bridges).
- `/rahome/bridge/sound/2` contains the same two format-specific subdirectories and their files. The appropriate subdirectory (based on your bridge type) is the source for Prompt Set 1.

If you want to add a second prompt set, you must create the directory `/rahome/bridge/sound/3`, and put the new `.wav` files into the appropriate format-specific subdirectory (`adpcm` or `g711`) for your system. The third set's files must be in the appropriate subdirectory of `/rahome/bridge/sound/4`, and so on.

Caution! When it responds to events by playing specific voice prompt files, the ReadiVoice system expects to find files with the specific names listed. When you create and install new prompt sets or replace prompts in existing prompt sets, be sure the files' names are correct. If the system can't find a file in the specific prompt set directory, it looks in the appropriate subdirectory of the default directory (/rahome/bridge/sound/1). Don't change or delete any of the files in this directory!

See "Voice Prompt File Reference" on page 182 for a list of the default voice prompt files, their contents, and their length.

When you install additional prompt sets, you must run the `tnl_mkln` script to create the appropriate symbolic links for the initial greetings. See "About Initial Greetings" on page 181 for a description of how symbolic links are used for greetings and what the `tnl_mkln` script does.

The **Prompt Sets** page in the System Administration interface lets you add new prompt sets to the ReadiVoice system, rename prompt sets, or delete prompt sets from the system (deleting a prompt set doesn't remove the files from the disk). To change the prompts themselves, you must replace the `.wav` files for the prompts you want to change.

For InnoVox 480 bridges, the `.wav` files must be in 4-bit IMA ADPCM format (8000 samples/second, 16-bit mono, compressed 4:1). For InnoVox 4000 bridges, they must be in 8-bit μ-law format (8000 samples/second, 16-bit mono or higher).

For the highest sound quality, prompt set files should be recorded in a professional sound studio. If you're upgrading from an InnoVox 480 bridge and want the same prompt set, create the μ-law encoded `.wav` files from the original recordings. Converting a 4-bit IMA ADPCM file into 8-bit μ-law won't provide noticeable quality improvements.

Adding a Prompt Set

To add a prompt set:

1 In the **System Administration** navigation bar, click **Prompt Sets**.

The **Prompt Set**s page appears (Figure 7-2), listing the available prompt sets. The **Path** column shows the directory path where the ReadiVoice system expects to find each prompt set's files.

The path shown doesn't include the format-specific subdirectory (`adpcm` or `g711`) that contains the actual files, but points to its parent directory.

2 In the empty field at the bottom of the **Name** column, type a name for the new prompt set. Then click the **Add** button.

The system confirms that the prompt set has been added and provides a link for returning to the **Prompt Sets** page.

3 Click the link to reload the **Prompt Sets** page.

The **Prompt Sets** page appears, with the new entry at the end of the list.

4 Make a note of the directory path assigned to the new set.

The remaining steps require you to have access to the CACS server or establish a Telnet session to it in order to install the files.

5 Use file transfer protocol (`ftp`) to copy the new prompt set's `.wav` files to a local directory on the CACS server. Be sure to use binary transfer mode.

6 On the CACS server, create the directory assigned to the new set. For instance, if this is the second prompt set, you'd create the directory `/rahome/bridge/sound/3.`

7 In the new prompt set directory, create the format-specific subdirectory needed for your system (`adpcm` or `g711`). If the new prompt set uses some of the same `.wav` files as an existing prompt set, it's easiest to copy this directory and its contents from the existing set.

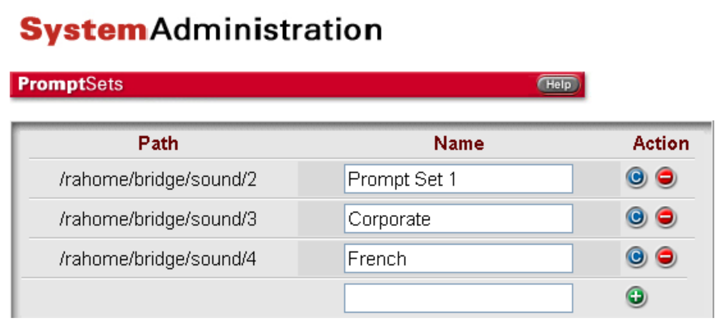
8 Copy or move the new `.wav` files for this prompt set from the temporary location where you put them in 5 into the subdirectory you created in 7, overwriting any existing files that you're replacing.

9 Repeat the preceding steps to add another prompt set.

10 When you're finished adding prompt sets, create the symbolic links for the initial greetings (see "About Initial Greetings" on page 181) by running the `tnl_mkln` script:

- **a** Log into the CACS as root and, in an XTerm window, enter **`tcsh`**.
- **b** Change to the `/rahome/bin` directory.
- **c** Run the make link script by entering **`tnl_mkln`**.
- **d** Reboot the bridge(s).

Figure 7-2 ***Prompt Sets** page*



Renaming a Prompt Set

To change the name of a prompt set:

1 In the navigation bar, click **Prompt Sets**.

The **Prompt Sets** page appears, listing the available prompt sets.

2 Edit the prompt set's name. Then click its **Commit** button (blue "c").

The system confirms that the prompt set has been modified.

Repeat the above procedure to rename another prompt set.

Deleting a Prompt Set

To remove a prompt set from your ReadiVoice system:

1 In the navigation bar, click **Prompt Sets**.

The **Prompt Sets** page appears, listing the available prompt sets.

2 Find the prompt to remove and click its **Delete** button (red "–").

The system confirms that the prompt set has been deleted.

Repeat the above procedure to delete another prompt set.

Deleting the prompt set from within System Administration only makes it unavailable to the ReadiVoice system. It doesn't delete the `.wav` files from the CACS server or remove the directory that contains them.

Customizing Greetings and Related Messages

In a ReadiVoice non-routed system and a ReadiVoice IP system, the bridge plays the initial greeting. You can configure ReadiVoice to use up to 50 custom greetings, so that it selects the appropriate greeting based on the *DNIS* (dialed number identification service) digits that the network delivers to the bridge.

Custom welcome messages aren't available in a routed (INCR) system because the network that routes calls to the bridges plays the initial greeting (welcome message) that callers hear. The carrier stores and plays this message.

The DNIS-specific initial greeting files you need depend on your system configuration:

- If your system is Non-routed Private, create DNIS-specific versions of the `helo_inbound.wav` file.
- If your system uses *only* the Traditional Shared call flow, create DNIS-specific versions of the `cr_access_code_prompt.wav` file.
- If your system *also* uses the Two-Password Shared call flow, create DNIS-specific versions of the `pw_prompt.wav` file.

When you run the `tnl_mkln` script, it creates the DNIS-specific symbolic links for the custom greeting files you created. See "About Initial Greetings" on page 181 for a description of how symbolic links are used for greetings and what the `tnl_mkln` script does.

In addition to the initial greeting, several related prompts can also be customized based on DNIS. The DNIS-customizable files are:

```
cf_call_not_completed
cf_conf_full_disconnect.wav
cr_access_code_collection_failure.wav
cr_access_code_good.wav
cr_access_code_prompt.wav
cr_access_code_wrong.wav
cr_access_code_wrong_.wav
_cr_access_code_wrong.wav
cr_conf_ending.wav
helo_inbound.wav
pw_prompt.wav
```

This feature is called *Welcome on DNIS*, and it works as follows:

1. During the call-setup protocol, the network delivers the call's DNIS number to the bridge.
2. The bridge looks in `/rahome/bridge/sound/greetings/adpcm` or `/rahome/bridge/sound/greetings/g711` (depending on bridge type) for a `.wav` file that starts with the DNIS digits received and matches the needed event.

For example, if the system is Non-routed Private, and the bridge receives the digits 1111, it plays `1111helo_inbound.wav` to the caller if the file exists. Another caller, whose DNIS digits are 2222, would hear `2222helo_inbound.wav` if the file existed.

3. If the DNIS-specific `.wav` file doesn't exist, the bridge plays the corresponding non-customized greeting (`helo_inbound.wav`, for example) in `/rahome/bridge/sound/1/adpcm` or `/rahome/bridge/sound/1/g711` (depending on bridge type).

To implement this feature:

1. Create the custom messages you want to use in the required format, and add the DNIS digits to the beginning of each standard file name.

   See Appendix A for descriptions of the customizable files (listed above), including the wording of the default versions. The appendix also describes the bridge-specific file formats.

   To use the same greeting (or other customized message) for several DNISes or access numbers, create multiple copies of the same `.wav` files, each named with its DNIS.

Caution! The custom file names must match exactly the DNIS digits delivered to the bridge. Typically, on a non-routed system, a DNIS is a four-digit number. If your system receives more DNIS digits, however, the file names must include all the digits.

2. Place the new appropriately-named files into the correct format-specific subdirectory (`adpcm` or `g711`) of the `/rahome/bridge/sound/greetings/` directory.

   ReadiVoice uses the default greeting-related files (from the appropriate subdirectory of `/rahome/bridge/sound/1`) for any DNIS that doesn't have its own `.wav` files. To use your own default greetings, replace the supplied default greeting files with ones of the same names containing the messages you want.

3. Create the DNIS-specific symbolic links (see "About Initial Greetings" on page 181) by running the `tnl_mkln` script:

   a. Log into the CACS as root and, in an XTerm window, enter **`tcsh`**.

   b. Change to the `/rahome/bin` directory.

   c. Run the make link script by entering **`tnl_mkln`**.

   d. Reboot the bridge(s).

4. Inform provisioners of the system of the specific access numbers that you've associated with custom welcome messages.

   If you've created corresponding custom prompt sets, give provisioners this information as well. See "Working with Prompt Sets" on page 166 for more information.

Creating Your Own Indexed WAVE Files

ReadiVoice uses an *indexed WAVE file* to store the short sound clips that are played in various combinations to speak numbers, dates, and times. Indexed WAVE files can also be used for other purposes.

Using the indexed WAVE file format, you can support several languages by supplying the necessary sound clips and building separate files for each language. The next section, "Building Indexed WAVE Files, describes how to create indexed WAVE files.

For more detailed information about WAVE files in general and indexed WAVE files in particular, see "Details About the WAVE File Format" on page 175 and "Indexed WAVE File Specifications" on page 176.

Building Indexed WAVE Files

Polycom's indexed WAVE file is the concatenation of several individual WAVE (`.wav`) files into a single file. Individual sound clips are accessed by using an index or offset into the file.

For InnoVox 480 bridges, both the source WAVE files and the output indexed WAVE file store the audio data using 4-bit IMA ADPCM format (8000 samples/second, 16-bit mono, compressed 4:1). For InnoVox 4000 bridges, both the source WAVE files and the output indexed WAVE file store the audio data using 8-bit µ-law format (8000 samples/second, 16-bit mono).

The two types of indexed WAVE files you can build are:

Predefined — contains sound clips for numbers, dates, times, etc. The number of sound clips in the file is language specific and may vary from language to language. Grammar rules for the language are applied during playback.

Generic — contains a fixed set of sound clips used to say words or phrases for the application. No grammar rules are applied during playback.

Both types of indexed WAVE files are built the same way, using the `genvoice` utility. You specify the key file to use, the directory location for the sound clips, and the output file name. Only setting up the key file is different.

Key File Guidelines

Follow these formatting guidelines when creating key files:

- Each entry must be on one line (carriage returns aren't legal white space).
- Lines containing a pound sign (**#**) in the first column are ignored (treated as comment lines).
- Blank lines are ignored.
- Identifiers can be separated by any amount of white space. White space is defined as tabs and spaces.

- *Only in predefined key files*, the first line contains `LANG=` followed by the upper-case English name of the desired language. These names are converted into an appropriate integer for the indexed WAVE file. Recognized languages are:
  - ENGLISH
  - GERMAN
  - JAPANESE
  - FRENCH
  - SPANISH
  - CHINESE
  - SWEDISH
- Each key entry on one line consists of:
  - An index number and a semicolon (`;`).
  - The entry name and a semicolon.
  - The name of the WAVE file.
- The index numbers in the predefined key files are fixed by the application. Don't change them.
- The name identifiers are for the benefit of the humans using the file, providing meaningful information about the entries. They aren't used by the application.
- If a WAVE file name includes a path, it's relative to the source file directory specified in the `genvoice` command (see page 174).

Setting Up a Predefined Key File

The key files for predefined indexed WAVE files look like this:

```
LANG=language
0; name_00; waveFile_00
   .
   .
   .
x; name_xx; waveFile_xx
```

Don't change the leading index numbers (*x*) or name identifiers (*name_xx*). When you change predefined key files, edit only the *waveFile_xx* fields so that they point to the correct sound files for the key entries. If necessary, include a path that's relative to the directory you specify in the `genvoice` command to build the indexed WAVE file (see "Running the Build Process" on page 174).

The example below shows part of a German predefined key file:

```
LANG=GERMAN
0; number_0; null.wav
1; number_1; eins.wav
2; number_2; zwei.wav
   .
   .
   .
107; 30th; dreizigste.wav
108; der; der.wav
109; ein; ein.wav
110; eine; eine.wav
```

Due to grammatical differences, the number of entries needed may vary from language to language. This is why Polycom supplies the predefined key files for the supported languages.

Creating a Generic Key File

An application can take advantage of indexed WAVE files for outgoing voice prompts. The format of the key files is identical to the format of predefined key files except that no language is specified.

You create a key file in order to build an application indexed WAVE file. It should look like this, without a line specifying the language:

```
0; name_00; waveFile_00
   .
   .
   .
x; name_xx; waveFile_xx
```

Your key file can contain any number of entries, and, of course, you can specify any name (*name_xx*) you like for each entry. As with the predefined key file, edit the *waveFile_xx* field for each entry to reference the correct sound file.

The example below shows part of a a generic key file:

```
0; welcome to the system;cf_you_subs_prompt.wav
1; howdy y'all; howdy.wav
   .
   .
   .
107; goodbye y'all; goodbye.wav
```

Because no grammar rules are employed during playback, the number of entries in generic key files can be the same from language to language.

Running the Build Process

To build an indexed WAVE file, use the `genvoice` utility in the `/rahome/bridge/binaries/tools/` directory as follows:

1. Create or edit the key file as described above.
2. Be sure that all the input WAVE files are located in the proper directory or directories.
3. At the command prompt, enter:

   `genvoice -i` ***`inputFile`*** **`-d`** ***`voiceDir`*** **`-o`** ***`outputFile`*** **`[-v]`**

`inputFile` specifies the key file to use. If the key file isn't in the current directory, *`inputFile`* must specify its path. InnoVox provides predefined key files for each language that it supports.

`voiceDir` specifies the directory where the source WAVE files are located. If the source files aren't in this directory, the entries in the key file must contain the relative path from this directory.

`outputFile` specifies the file name for the indexed WAVE file being created. If *`outputFile`* doesn't specify a path, the file is created in the current directory.

`-v` specifies that the WAVE file is generated with µ-law encoding. If `-v` isn't specified, IMA ADPCM encoding is used.

InnoVox 480 bridges require IMA ADPCM encoded WAVE files. InnoVox 4000 bridges require µ-law encoded WAVE files. If you're upgrading from an InnoVox 480 bridge and want the same prompt set, you should create the µ-law encoded WAVE files from the original recordings at 8000 samples/sec., 16-bit mono (or higher quality). Converting a 4-bit IMA ADPCM into 8-bit µ-law doesn't provide noticeable quality improvements.

The `genvoice` utility displays an appropriate error message and terminates if:

- A source file is in the wrong format or can't be found.
- The key file is in the wrong format or can't be found.
- The number of entries found in the predefined key file doesn't match the expected number for the specified language. This check doesn't apply to generic key files.

An indexed WAVE file can be played using any program that supports WAVE, but the indices will be ignored. Some sound editors may not be able to play the file correctly because of the compression used.

Details About the WAVE File Format

A WAVE file stores digital (sampled) audio data in Microsoft's *Resource Interchange File Format (RIFF)*. RIFF is a variant of Electronic Arts' *Interchange File Format (IFF).* Unlike IFF, the Intel-oriented RIFF uses Little Endian byte order. For more information about RIFF, visit the Microsoft Developer Network (www.msdn.microsoft.com).

RIFF files consist of logical units of data called *chunks*. The first chunk in a RIFF file is the RIFF chunk. All other chunks in the file are *subchunks* of the RIFF chunk.

At a minimum, all WAVE files contain:

- The top-level RIFF chunk, which consists of three parts:
  - The chunk identifier, which is the four-character code `RIFF`.
  - The size of the data member chunks, a `DWORD` (four-byte integer). The size is everything after the first eight bytes (the chunk identifier and the size `DWORD` itself).
  - The data member, which begins with a `DWORD` that specifies the form type of the chunk (in this case `WAVE`). All other chunks in the file are considered part of the data member of the RIFF chunk.
- A Format subchunk that describes the data (sample rate, bit resolution, type of compression, and so forth).
- A Data subchunk that contains the actual sampled audio data.

But, RIFF (like IFF) is extensible. WAVE files can contain, and developers can define, additional types of chunks to provide additional data and functionality.

Polycom added two new chunks to the WAVE file format to create its indexed WAVE files. They're described in the "Indexed WAVE File Specifications" section.

Indexed WAVE File Specifications

In Polycom's indexed WAVE files, the Data chunk consists of a set of individual sound files that are concatenated together. An Index chunk enables Polycom applications to locate the individual sound clips and quickly piece them together to play phrases, numbers, dates, and so forth. The sound clips can be common words, digits, alphabetic letters, or even short phrases.

An indexed WAVE file has:

- The chunk identifier, which is the four-character code `RIFF`.
- The size of the data member chunk, a `DWORD` (4-byte integer). The size is everything after the first eight bytes (the chunk identifier and the size `DWORD` itself).
- The data member is the largest portion of the file. It consists of:
  - A `DWORD` that specifies the form type of the chunk (in this case `WAVE`).
  - A Format subchunk that describes the data (sample rate, bit resolution, type of compression, and so forth).
  - A Language chunk identifying the language of the sound clips.
  - An Index chunk containing all of the offset information. The Index chunk contains:
    - » The size (`DWORD`) of the Index chunk.
    - » The number (`DWORD`) of sound clips in the WAVE file.
    - » The offset location (`DWORD`) and length in bytes (`DWORD`) of each sound clip in the file.
  - The Data subchunk contains the actual sampled audio data, or sound clips. These are the data subchunks pulled out of individual WAVE files and concatenated here.

The specifications for these chunks use Microsoft's RIFF notation. Look up RIFF notation on the Microsoft Developer Network (www.msdn.microsoft.com).

Language Chunk

The Language chunk identifies the language associated with the indexed WAVE file. This lets applications using the file apply the appropriate grammar, where required, for building number or date phrases correctly in that language. An indexed WAVE file can have only one language identifier.

Only predefined indexed WAVE files use the language field. You specify other languages using the appropriate predefined key file. Supported languages are:

- CHINESE
- ENGLISH
- FRENCH

- GERMAN
- JAPANESE
- SPANISH
- SWEDISH

The Language chunk consists of the chunk identifier (`lang`), the size of the data member (`DWORD`), and the two-byte integer (`WORD`) language field. In RIFF notation:

```
<lang-ck> ->
lang(                  // chunk ID
   <size:DWORD>        // data member size
   <language:WORD>)    // language
```

Index Chunk

The Index chunk identifies the starting position (offset) and length of each segment (sound clip) in the file. The chunk begins with the chunk identifier `bmak` (for “bridge multi-access key”). This is followed by a 4-byte integer (`DWORD`) containing the length in bytes of the Index chunk’s data member (that is, the size of the chunk, not including the chunk ID and size fields).

The data member itself begins with a 4-byte integer (`DWORD`) specifying the number of index entries. This is followed by the `indexPair` struct containing the actual index data. In RIFF notation:

```
<bmak-ck> ->
bmak(                  // chunk ID
   <size:DWORD>        // size of index chunk
   <numIdxPairs:DWORD> // number of index pairs
   <indexPair> …)      // index pair list

<indexPair> ->
struct
   {
      DWORD offset;     // Absolute offset of
                        // segment, in bytes.
      DWORD length;     // Length of segment, in bytes.
   }
```

6

Diagnostics and Troubleshooting

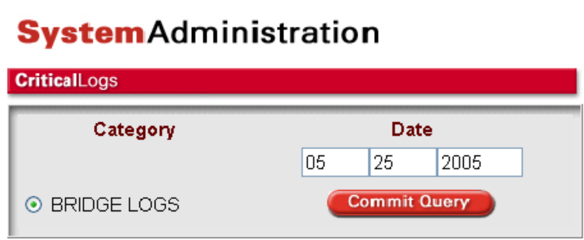
This chapter describes how to perform some diagnostic and troubleshooting tasks.

Viewing Critical Logs

The **Critical Log**s page (Figure 6-1) appears when you click the **View Critical Logs** link in the ReadiVoice home page's navigation bar. It gives you access to critical troubleshooting information that may require corrective action.

Processing the log files imposes significant processor and memory burdens on your computer and can take some time. Log files are typically useful only to trained technicians, and even trained technicians typically need assistance from Polycom Global Services.

Figure 6-1 ***Critical Logs** page*



On this page, select **BRIDGE LOGS** and a date. and then click **Commit Query**. ReadiVoice searches the raw log files for the date you entered and retrieves critical entries.

When the system is operating normally, the critical logs should be nearly empty. If the system is in a testing mode or other abnormal state that generates large raw log files, your query may take a very long time.

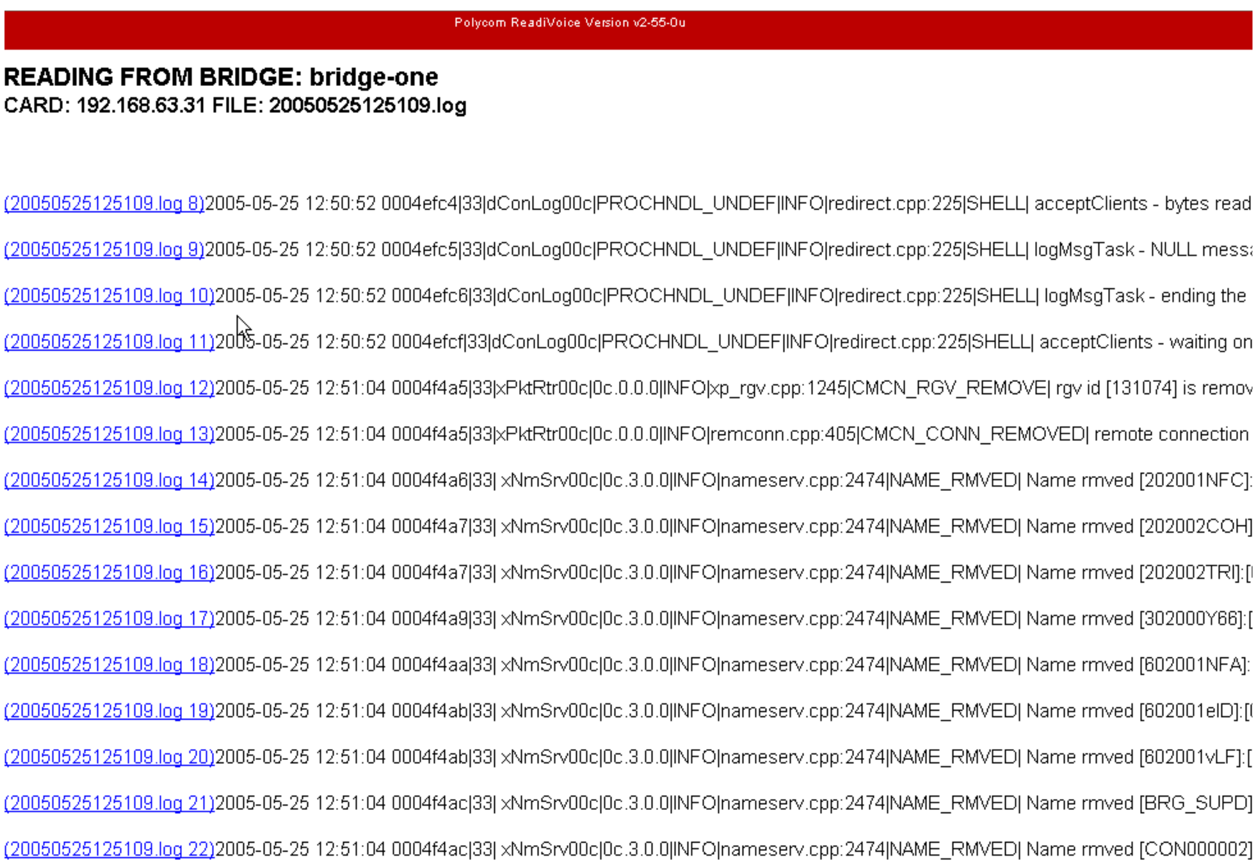
The critical log file shows you critical entries about bridge processes, such as:

- Bridge up or down.
- Span trouble (doesn't apply to IP systems).
- Port capacity changes.
- Bridge clock source changes or failures.
- Abnormal bridge task deletion or suspension.

Figure 6-2 shows a bridge critical log page. Typically, a properly running system has few or no entries.

Each critical log entry contains a link to the raw log file from which it came. To investigate an entry further, click its link. The system retrieves a portion of the raw log file that shows the entry in context (from 100 lines before to 100 lines after the entry).

Figure 6-2 *Bridge critical log page*



Database Troubleshooting

This section suggests some methods for diagnosing the database and investigating database problems.

Checking for Informix Server Engine Errors

To determine if an error has occurred in the Informix server engine, check the `/usr/informix/online.log` file for `error` or `panic` messages, using the `grep` command to search for those words.

With the standard configuration, Informix temporarily stores results data in the `/tmp` directory. Lack of sufficient free space in this directory can be the cause of server engine errors.

Use the `df` command to check the disk space in the `/tmp` directory (see "Checking Solaris Disk Space" on page 103). If it's more than 90% full, remove unneeded files to free up space for database data. Call your Polycom Global Services representative if you need help in determining what you can safely remove.

Checking for Slow SQL Queries

If you suspect the database is slowing down the system, check for an SQL query that takes much more time than the others:

1. In a CACS XTerm window, log in as root, switch user to informix, and set the Informix environment by entering **`tcsh`**.
2. Enter **`onstat -g ses`**.

   Informix displays a list of sessions and their memory usage.
3. Locate the session that uses the most memory and note its session ID.
4. Enter **`onstat -g sql`**.

   Informix displays a list of active queries and their session IDs.
5. Check the list of active queries for the session you identified in 3. If it's listed, enter **`onstat -g sql`** ***`ID`*** (where ***`ID`*** is the session ID number of the session you identified in 3).

   Informix displays the running query.
6. Contact your Polycom Global Services representative to determine if this query is taking more memory than it should.

Logging Database Activity with the Database Monitoring Script

Your ReadiVoice installation includes a Perl script, `dbMon.pl`, that logs Informix database information (such as number of active sessions and amount of memory used per session) into a file that can be used later for analysis.

If your system seems to be slowed by SQL queries that are taking inordinately long, it may be helpful to run this script and provide the resulting log file to your Polycom Global Services representative.

Caution! Use this script only if you are trained and knowledgeable in working with Solaris, Informix, and the ReadiVoice system.

The `dbMon.pl` script:

- Resides in the `/rahome/database/scripts` directory.
- Must be run by root.
- Should normally be run in the background.

The script creates a log file named *`YYYYMMDDhhmmss`*`dbMon.log`, where *`YYYYMMDDhhmmss`* is the system date and time (using a 4-digit year and 24-hour time) when the script is run. For example, if you run `dbMon.pl` on August 14th, 2002, at 2:22:37 PM, it creates a log file named `20020814142237dbMon.log`.

The log files are written to the `/rahome/database/scripts/archives/` directory.

Running the dbMon.pl Script

The syntax for running the `dbMon.pl` script is:

```
dbMon [-r n &]
```

To use the runtime option (`-r`), replace *`n`* with the number of minutes you want the script to run. If you omit this option, the script runs for the default duration of 5 minutes. Use `&` to make it a background process.

For example, enter **`dbMon.pl -r 20 &`** at the command prompt to run the script in the background for 20 minutes.

Contents of dbMon.pl Log File

The `dbMon.pl` script writes data to the log file at one-second and ten-second intervals.

Logged every second:

Active SQL statements: This section shows the session IDs and the statement types (`SELECT`, `UPDATE`, `INSERT` or `UPDATE STAT`) of the SQL statements that are currently running.

Active sessions: This section shows the session IDs, user types (such as root, web, or cnow), and total used memory of active database connections.

With this information, you can identify queries that are being run constantly against the database and determine the approximate time (to the nearest second) that it takes to run these queries.

Logged every ten seconds:

Number of reads/writes per user thread: This section shows the number of read and write operations performed on the database by each thread.

Processes waiting on latches, locks and transactions: This section shows the process IDs of processes waiting on latches, locks, or transactions.

With this information, you can identify the most active session (based on the memory usage) and the processes that are being suspended or waiting on locks or latches.

Telephony Troubleshooting

Your ReadiVoice system includes two ways of testing the function of the bridges:

- Dial-out testing, using the telephony test utility, `diagnose`.
- Dial-in testing, using the `LOG_CHAN_DEBUG` parameter in the `ive.ini` file.

These procedures are normally used during installation or upgrades, but may be run as needed, as long as the affected bridges are not being used for live conferencing.

Dial-Out Channel Testing

The `diagnose` telephony test utility is run from the command line. You can specify these parameters:

- The ID of the bridge you want to test.
- The equipment on the specified bridge that you want to test as:
  - Any *range* of cards, spans, and channels.
  - A *specific* card, span, and channel.
  - Any combination of ranges and specific cards, channels, or spans.

  In an IP system, there are no spans or channels as such. You can only specify the cards that you want to test.
- The phone number you want the bridge to call to test the equipment you specified.

When you run the `diagnose` utility:

1. The bridge you're testing calls you at the number you specified, using the first bridge resource to be tested.
2. When you answer the call, the bridge plays three numbers that identify the card, span, and channel on which the call is being made.

   In an IP system, the bridge plays a single number indicating the card on which the call is being made.
3. When you hang up, the bridge calls you on the next resource to be tested and repeats the process until it completes the test.
4. For each test call completed, the utility writes a line to a log file.

   The log file is specified in the `diagnosticlogfile` parameter of the `.odprocrc` file. The default setting is:

   ```
   diagnosticlogfile = /rahome/bin/TTU.log
   ```

   If this file exists, the utility appends the new test call records to it. Otherwise, it creates the file, adds a heading, and writes the records.

To start dial-out testing:

1. Open a Telnet session to the CACS as user cnow, and switch user to root.
2. Change to the `/rahome/utils` directory.
3. Type **`./diagnose`** and press ENTER.

   You see:

```
 To quit, press 'q':
 To Go To Previous Step, Press 'p':

Enter the ID of the Bridge to Test:
```

4. Type the ID of the bridge and press ENTER. To find bridge IDs, go to the **Bridges** page in System Administration (see page 18).

   You see:

```
Enter the Phone Number to Dial Out To:
```

5. Enter the telephone number at which you want the bridge to call you exactly as it should be dialed. Type only the digits, with no parentheses or dashes, and then press ENTER.

   You see:

```
Enter Range of Cards:
From:
```

6 Enter the logical number of the first card you want to test (card numbering begins with zero).

You see:

```
Enter Range of Cards:
To:
```

7 Enter the logical number of the last card you want to test. To test only the card you entered in step 6, enter the same number again.

If this is an IP system, The script displays a blank line, and your phone rings. Skip to step 12.

If this is a PSTN system, you see:

```
Press -1- To Test All Spans On A Board Or -2- For Individual
```

8 Choose one of these options:

– To test both spans on the selected card or cards, enter **1**. Then skip to step 10.

– To test a single span, enter **2**.

You see:

```
Enter the Span Port Number:
```

9 Enter the number of the span you want to test (**0** or **1**).

You see:

```
Press -1- To Test All Channels On A Span Or -2- For A Particular
Channel:
```

10 Choose between these options:

– To test all channels on the span, enter **1**. Then skip to step 12.

– To test a single channel, enter **2**.

You see:

```
Enter the Channel:
```

11 Enter the number of the channel you want to test.

The script displays a blank line, and your phone rings.

12 Answer the phone, listen to the message, and then hang up.

For each call, you hear three numbers indicating the card, span, and channel being used for the call (or, in an IP system, a single number indicating the card).

The details of each test call are recorded in the log file, so you need only note a channel if it has audio problems, such as static.

13 Choose one of these options:

- Continue answering the phone each time it rings, listening to the message and then hanging up, until the bridge stops calling you.
- To stop the dial-out test process before it completes, type CTRL + C. This also stops the updating of the diagnostic log file.

14 View the diagnostic log file (by default, `/rahome/bin/TTU.log`).

A line for each test call shows the card, channel, and span numbers, and the outcome of the call. For a successful test of channel 13 on span 0, card 3 of the bridge you tested, the log file would contain the line:

```
3/0/13    SUCCESS
```

In an IP system, the line for each test call shows the bridge and card numbers and the outcome of the call. For a successful test of card 0 of bridge 1, the log file would contain the line:

```
1/0    SUCCESS
```

Caution! If you find telephony problems, such as any outcome other than “SUCCESS,” print the file for your records or save the file under a new name or in a safe location. To enable Polycom Global Services to help you, be sure have this file available to send us.

Dial-In Channel Testing

A dial-in channel test script runs automatically when you set the `LOG_CHAN_DEBUG` parameter in the `ive.ini` file to `1`. By default, this parameter is set to `0`.

Caution! Don’t enable dial-in channel testing on a live conferencing bridge or bridges. When dial-in testing is turned on, callers hear a debug message (stating the card, span, and channel numbers—card only for IP—on which their call has reached the bridge) before the standard welcome message.

To start dial-in channel testing:

1. Remove the bridge or bridges from service and wait until all conferencing activity has finished.
2. Open the `ive.ini` file for editing.
3. In the `[Debug]` section, set `LOG_CHAN_DEBUG` to `1`.

   The edited line should read:

   ```
   LOG_CHAN_DEBUG = 1
   ```
4. Save and close the edited `ive.ini` file.
5. Reboot the bridge or bridges so that the processors reread the `ive.ini` file and begin running in channel-debug mode.
6. Call the ReadiVoice system and listen for the board, span, and channel numbers to be played before the welcome message (or, in an IP system, a single number indicating the card).
7. As you use the system, make a note of the specific channels (or cards), if any, associated with static or other audio problems.
8. When you're finished testing and want to make the system available for live conferencing, reopen the `ive.ini` file that you modified and restore the `LOG_CHAN_DEBUG` setting to `0`.

   The edited line should read:

   ```
   LOG_CHAN_DEBUG = 0
   ```
9. Save and close the edited `ive.ini` file.
10. Reboot the bridge or bridges so that the processors reread the `ive.ini` file and stop running in channel-debug mode.
11. Call the ReadiVoice system and verify that the standard welcome message plays *without* a debug message preceding it.

The system can now be restored to normal conferencing service.

A

Voice Prompts and Call Flows

This appendix describes the default voice prompts installed with the ReadiVoice system, how they're used, and the ReadiVoice call flows.

Overview

You can use multiple sets of voice prompts in the ReadiVoice system, which you can customize for specific needs. Before you start customizing your voice prompts or installing additional voice prompt sets, it helps to familiarize yourself with the default voice prompt set.

The installation process puts the default voice prompt files into the following locations:

- `/rahome/bridge/sound/1/adpcm` contains the `.wav` files in ADPCM format (for InnoVox 480 bridges).
- `/rahome/bridge/sound/1/g711` contains the `.wav` files in μ-law format (for InnoVox 4000 bridges).
- `/rahome/bridge/sound/2` contains the same two format-specific subdirectories and their files. The appropriate subdirectory (based on your bridge type) is the source for Prompt Set 1.

If you want to add a second prompt set, you must create the directory `/rahome/bridge/sound/3`, and put the new `.wav` files into the appropriate format-specific subdirectory (`adpcm` or `g711`) for your system. The third set's files must be in the appropriate subdirectory of `/rahome/bridge/sound/4`, and so on.

The **Prompt Sets** page in the Administration interface lets you add new prompt sets to the ReadiVoice database (making them available for provisioning), rename prompt sets, or delete prompt sets from the ReadiVoice database (making them unavailable, but not removing the files themselves). See "Working with Prompt Sets" on page 166.

To change the prompts themselves, you must replace the `.wav` files for the prompts you want to change. Use the information in this appendix to help you determine the appropriate text for your customized messages.

The default prompt set includes a file, `silence.wav`, containing only a very brief silence. If you want to replace a specific prompt with nothing (that is, you don't want any message to play in the situation where that prompt is called), you can replace it with a copy of `silence.wav` renamed to replace the `.wav` file that you don't want played.

The format of the `.wav` files depends on the type of bridges in your system:

InnoVox 480 bridges: All `.wav` files must be in 4-bit IMA ADPCM format (8000 samples/second, 16-bit mono, compressed 4:1). If your sound editor application offers multiple format options, use the DVI option when saving.

InnoVox 4000 bridges: All `.wav` files must be in 8-bit µ-law format (8000 samples/second, 16-bit mono). If you're upgrading from an InnoVox 480 bridge and want the same prompt set, you should create the µ-law encoded `.wav` files from the original recordings at 8000 samples/sec., 16-bit mono (or higher quality). Converting a 4-bit IMA ADPCM into 8-bit µ-law won't provide noticeable quality improvements.

You can use the `file` command to determine whether a `.wav` file is in ADPCM or µ-law format. Figure A-1 shows an example of checking two `.wav` files.

Figure A-1 *Checking the format of `.wav` files*

```
jaba </rahome> cnow> file bridge/sound/1/adpcm/0.wav
bridge/sound/1/adpcm/0.wav:     WAVE file, ADPCM format (mono)
jaba </rahome> cnow> file bridge/sound/1/g711/0.wav
bridge/sound/1/g711/0.wav:      WAVE file, G711 format (mono)
jaba </rahome> cnow>
```

About Initial Greetings

In a non-routed system, the bridge plays the initial greeting to callers. The file that's appropriate depends on your system configuration:

- If your system is Non-routed Private, then the initial greeting is always the helo_inbound.wav file, which doesn't prompt the caller to enter anything.
- If your system is Non-routed Shared, then, depending on the call flow, the caller could have either an access code (Traditional Shared) or a subscriber or participant password (Two-Password Shared). And the system may have both those call flows enabled.

The default prompt set includes two greeting files with different prompts:

- `cr_access_code_prompt.wav` specifically asks for an access code. It's appropriate if *only* the Traditional Shared call flow is enabled.
- `pw_prompt.wav` is more generic, asking for a "passcode." It's appropriate if *both* Traditional Shared and Two-Password Shared are enabled, since the system doesn't know initially which call flow applies to a caller.

The bridge doesn't actually look for either of those files. Instead, it looks for the `cr_code_prompt.wav` file, which is a symbolic link to one of the above.

During system installation, the `tnl_mkln` script must be run. Among other things, it sets up `cr_code_prompt.wav` as a symbolic link to the appropriate prompt, depending on system configuration:

- If your system uses only the Traditional Shared call flow, the symbolic link points to the `cr_access_code_prompt.wav` file.
- If your system also uses the Two-Password Shared call flow, the symbolic link points to the `pw_prompt.wav` file.

If you ever change those call flow settings, add prompt sets, or set up custom greetings, you must run the `tnl_mkln` script again so that it can change the symbolic links, if necessary, or add them for the new prompt sets or custom greetings.

In the case of custom greetings (see "Customizing Greetings and Related Messages" on page 169), the `tnl_mkln` script creates a symbolic link for each DNIS for which it finds the target file.

For instance, if the `/rahome/bridge/sound/greetings/adpcm` directory contains a file named `1234gencode_prompt.wav` (and the Two-Password Shared call flow is enabled), the `tnl_mkln` script creates a symbolic link to it named `1234pass_access_code.wav`.

Voice Prompt File Reference

This section documents the default voice prompt files installed in the ReadiVoice system. Table A-1 lists the default voice prompt files in alphabetical order.Use it as a reference when contemplating new voice prompt sets and to determine which voice prompts may need to be customized.

Caution! When it responds to events by playing specific voice prompt files, ReadiVoice expects to find files with the filenames listed. When you create and install new prompt sets or replace prompts in existing prompt sets, be sure the files' names are correct.

Table A-1 *All voice prompts, alphabetized*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`0.wav`	"Zero"	All	0.4
`1.wav`	"One"	All	0.5
`10.wav`	"Ten"	All	0.6
`11.wav`	"Eleven"	All	0.7
`12.wav`	"Twelve"	All	0.5
`13.wav`	"Thirteen"	All	0.8
`14.wav`	"Fourteen"	All	0.7
`15.wav`	"Fifteen"	All	0.8
`16.wav`	"Sixteen"	All	0.9
`17.wav`	"Seventeen"	All	1.0
`18.wav`	"Eighteen"	All	0.8
`19.wav`	"Nineteen"	All	1.0
`2.wav`	"Two"	All	0.4
`20.wav`	"Twenty"	All	0.6
`3.wav`	"Three"	All	0.4
`30.wav`	"Thirty"	All	0.6
`4.wav`	"Four"	All	0.5

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`40.wav`	“Forty”	All	0.5
`5.wav`	“Five”	All	0.6
`50.wav`	“Fifty”	All	0.6
`6.wav`	“Six”	All	0.4
`60.wav`	“Sixty”	All	0.5
`7.wav`	“Seven”	All	0.6
`70.wav`	“Seventy”	All	0.6
`8.wav`	“Eight”	All	0.5
`80.wav`	“Eighty”	All	0.5
`9.wav`	“Nine”	All	0.8
`90.wav`	“Ninety”	All	0.7
`ao_change_failure.wav`	“I’m sorry, the system is currently unable to change this option, please try again later.”	Account Options menu	6.4
`ao_conf_return_prompt.wav`	“To return to conference, press star.”	Announcement setting process	1.8
`ao_failure.wav`	“I’m sorry, the system is currently unable to change this option, please try again later.”	Account Options menu	6.4
`ao_feature_overview.wav`	“Account Options feature overview…”	Account Options menu	2.7
`ao_initial_menu.wav`	“To start or join your conference, press 1. To change Account Options, press 2.”	Account Options menu	7.0
`ao_main_menu_.wav`	“Account Options. Changes to your account options affect both the current conference and future conferences, except that quick start and listen only changes affect only future conferences. You may change your account options at any time by returning to this menu.”	Account Options menu	
`_ao_main_menu.wav`	“For an overview of Account Options, press 9.”	Account Options menu	2.5

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`ao_none_avail.wav`	"This option is not currently available."	Account Options process	
`ao_not_enabled.wav`	"I'm sorry, this feature is not enabled."	Account Options menu	2.5
`ao_press_1.wav`	"Press 1."	Account Options menu	1.2
`ao_press_2.wav`	"Press 2."	Account Options menu	1.3
`ao_press_3.wav`	"Press 3."	Account Options menu	1.3
`ao_press_4.wav`	"Press 4."	Account Options menu	1.3
`ao_press_5.wav`	"Press 5."	Account Options menu	
`ao_press_6.wav`	"Press 6."	Account Options menu	
`ao_press_7.wav`	"Press 7."	Account Options menu	
`ao_press_8.wav`	"Press 8."	Account Options menu	
`ao_previous_menu_prompt.wav`	"To return to the previous menu, press star."	Account Options menu	3.7
`ao_wrong_key.wav`	"I'm sorry, that entry is not valid."	Account Options menu	3.0
`cc_active_reminder.wav`	"The conference will be allowed to continue after you disconnect."	Continuation [Sub call flow]	3.7
`cc_auto_change_prompt_.wav`	"To change the Auto Continuation option…"	Continuation [Autocontinuation setting]	3.0
`_cc_auto_change_prompt.wav`	<silence>	Continuation [Autocontinuation setting]	
`cc_auto_off.wav`	"Auto Continuation is off."	Continuation [Autocontinuation setting]	2.9

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`cc_auto_on.wav`	"Auto Continuation is on."	Continuation [Autocontinuation setting]	2.9
`cc_auto_overview.wav`	"Auto Continuation turns continuation on for all of your conferences, so they can continue after you disconnect."	Continuation [Account Options help]	
`cc_auto_set_off_prompt.wav`	"To turn Auto Continuation off, press 1."	Continuation [Autocontinuation setting]	4.5
`cc_auto_set_on_prompt.wav`	"To turn Auto Continuation on, press 1."	Continuation [Autocontinuation setting]	4.5
`cc_change_not_enabled.wav`	"I'm sorry, this feature is not enabled."	All [In conf sub]	2.5
`cc_conf_end_disconnect.wav`	"The subscriber has disconnected. The conference will now end."	Continuation off	4.1
`cc_off.wav`	"The conference will end when you disconnect. To allow the conference to continue after you disconnect, press star 8."	Continuation available [In conf sub]	8.3
`cc_on.wav`	"The conference will be allowed to continue after you disconnect. To set the conference to end when you disconnect, press star 8."	Continuation [In conf sub]	8.9
`cf_call_not_completed.wav`	"I'm sorry, your call could not be completed. Please try your call again."	All [Dial-in]	4.2
`cf_caller_count_.wav`	"There are…"	Conference Entry with Count [Sub call flow] [Part call flow]	
`_cf_caller_count.wav`	"…participants in the conference."	Conference Entry with Count [Sub call flow] [Part call flow]	
`cf_conf_full_disconnect.wav`	"I'm sorry, the conference you are attempting to join is full."	All [Sub call flow] [Part call flow]	3.6

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`cf_conf_start.wav`	"Thank you. Your conference will now begin."	All [Sub call flow]	2.5
`cf_default_non_part_disconnect.wav`	"Thank you."	All (default part disconnect message)	0.7
`cf_default_part_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	All (default non-part disconnect message)	4.4
`cf_eip_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	All [Sub call flow] [Part call flow]	4.4
`cf_first_caller.wav`	"You are the first participant to join the conference, please stand by."	Quick Start [Part call flow]	4.6
`cf_in_conf_help_cmd.wav`	"For a menu of available commands, press star star at any time during the conference."	All [Sub call flow]	6.1
`cf_in_conf_part_cmd_help.wav`	"The following conference commands are available: To request an Operator, press star zero. To request an Operator join your conference, press zero zero. To mute your line, press star 6. To un-mute your line, press star 7. To hear a list of conference participants, press star 9. Participant count, press star pound."	All [Part - in-conf dtmf]	22.0

***Table A-1** All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`cf_in_conf_subs_cmd_help.wav`	"The following conference commands are available to the subscriber: To request an operator, press star zero. To request an operator join your conference, press zero zero. To dial out, press star 1. To add a recorder to conference, press star 2. To change the conference entry and exit announcement options, press star 3. To lock your conference, press star 4. To unlock your conference, press star 5. To mute your line, press star 6. To unmute your line, press star 7. To allow the conference to continue after you disconnect, press star 8. To hear a list of conference participants, press star 9. To hear participant count, press star pound. To mute all but the subscriber, press pound pound. To unmute all, press 99. To place all callers in Listen Only except the subscriber, press pound 1. To turn off Listen Only, press pound 2. To process callers in the Waiting Room, press pound 5. To toggle the waiting room notifications announcement setting on or off, press pound 6."	All [Sub - in-conf dtmf]	
`cf_late_subscriber.wav`	"Someone has already joined this conference using the subscriber password. You'll join the conference as a participant."	All [Part call flow]	
`cf_music.wav`	<hold music>	All [Part call flow]	30.0
`cf_music_hold_to_conf.wav`	"The subscriber has joined the conference. Your conference will now begin. Please stand by."	Large conference music hold extender [Part call flow]	
`cf_non_part_join.wav`	"You will now be placed into conference."	All [Sub call flow]	2.4
`cf_part_join.wav`	"You will now be placed into conference."	All [Part call flow]	2.4

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`cf_part_music_wait.wav`	"The subscriber has not yet arrived. Please stand by."	Quick Start off [Part call flow]	4.3
`cf_please_standby.wav`	"Please wait."	All [Newcomer, dial-in or dial-out]	
`cf_subs_disconnect.wav`	"You have been disconnected by the conference moderator. Good-bye."	Dial-out [Dial-out call flow]	3.7
`cf_subs_first_caller.wav`	"You are the first participant to join the conference, please stand by."	All [Sub call flow]	4.6
`cf_you_subs_prompt.wav`	"The subscriber has not yet arrived. If you are the subscriber, please press the star key now."	Prompt for subscriber [Dial-in, trad. shared]	6.6
`cr_access_code_collection_failure.wav`	"I'm sorry, your call could not be completed. Please try your call again."	Shared Access System [Dial-in, shared]	4.2
`cr_access_code_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	Shared Access System [Dial-in,shared]	4.4
`cr_access_code_good.wav`	"Thank you."	Shared Access System [Dial-in, shared]	0.7
`cr_access_code_prompt.wav`	"Welcome to the Conferencing Center. Please enter your access code followed by the pound sign."	Shared Access System [Dial-in, trad. shared]	4.6
`cr_access_code_wrong.wav`	"I'm sorry, that entry is not valid."	Shared Access System [Dial-in, shared]	
`cr_access_code_wrong_.wav`	"I'm sorry, your entry…"	Shared Access System [Dial-in, shared]	1.9
`_cr_access_code_wrong.wav`	"… is not valid. Please enter the valid digits followed by the pound sign."	Shared Access System [Dial-in, shared]	5.2
`cr_access_number_wrong.wav`	"I'm sorry. This access number is not correct."	Private Access System	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`cr_code_prompt.wav`	Not a prompt, but a symbolic link to either `cr_access_code_prompt.wav` or `pw_prompt.wav`. See "About Initial Greetings" on page 181.	Shared Access System [Dial-in, shared]	
`cr_conf_ending.wav`	"This conference is not currently available. Please try your call again later."	All but routed and IP [Dialin]	5.0
`cr_conf_full.wav`	"I'm sorry, the conference you are attempting to join is full."	All but routed and IP [Dialin]	3.6
`cr_system_busy.wav`	"This conference is not currently available. Please try your call again later."	All but routed and IP [Dialin]	5.0
`csn_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	Conference Security Code [CSC setting]	4.4
`csn_mandatory_setup_prompt.wav`	"You must set up a conference security code. Please enter a 4 to 9 digit code followed by the pound sign."	Conference Security Code [CSC setting]	9.5
`csn_none_set.wav`	"A conference security code will not be required for this conference."	Conference Security Code [CSC setting]	4.0
`csn_prompt.wav`	"Please enter the conference security code followed by the pound sign."	Conference Security Code [Part CSC flow]	3.5
`csn_setup_prompt.wav`	"To bypass setting up a conference security code press star now. To set up a conference security code please enter a 4 to 9 digit code followed by the pound sign."	Conference Security Code [CSC setting]	13.6
`csn_verify_prompt_.wav`	"Your conference security code is…"	Conference Security Code [CSC setting]	2.0
`_csn_verify_prompt.wav`	"…To change this entry, press star now."	Conference Security Code [CSC setting]	2.8
`csn_wrong.wav`	"I'm sorry, that entry is not valid."	Conference Security Code [Part CSC flow]	3.0

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
csn_wrong_.wav	"I'm sorry, your entry…"	Conference Security Code [Part CSC flow]	1.9
_csn_wrong.wav	"… is not valid. Please enter the valid digits followed by the pound sign."	Conference Security Code [Part CSC flow]	5.2
csn_wrong_length.wav	"I'm sorry, the conference security code must be 4 to 9 digits long."	Conference Security Code [CSC setting]	8.6
ct_auto_disconnect.wav	"This conference has exceeded the time allowed for a single-participant conference. You will now be disconnected. Goodbye."	Conference Termination	
ct_auto_prompt.wav	"Are you there? If you wish to remain in conference, please press any key now."	Conference Termination	
ct_conf_timeout_ disconnect.wav	"The subscriber has not arrived. Please try your call again later."	Quick Start off [Music hold]	4.4
ct_manual_canceled.wav	"The conference will continue."	All [Sub end conf]	
ct_manual_disconnect.wav	"The subscriber has ended the conference. Goodbye."	All [Sub end conf]	
ct_manual_end.wav	"The conference has ended. All participants will be disconnected."	All [Sub end conf]	
ct_manual_prompt.wav	"To end this conference, press 1; to continue this conference, press 2."	All [Sub end conf]	
do_auto_disconnect.wav	"You have been disconnected by the system. Please try your call again."	Dial-out [Dial-out part call flow]	
do_auto_join_failure.wav	"The dial out was unsuccessful."	Dial-out [Dial-out part call flow]	
do_auto_join_prompt.wav	"You are being called to join a teleconference. Press 1 to join."	Dial-out [Dial-out part call flow]	4.5
do_bad_num_.wav	<silence>	Dial-out [Sub dial-out]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`_do_bad_num.wav`	"… is not valid. Please enter the vaild digits followed by the pound sign."	Dial-out [Sub dial-out]	1.8
`do_blast_dial_subname_announce_.wav`	<silence>	Dial-out [Dial-out part call flow]	
`_do_blast_dial_subname_announce.wav`	"... is calling you to join a teleconference. Please press one to join."	Dial-out [Dial-out part call flow]	
`do_conf_full.wav`	"No additional people can be added to this call."	Dial-out [Sub dial-out]	3.6
`do_dialing.wav`	"Dialing."	Dial-out [Sub dial-out]	0.8
`do_extplay_.wav`	"Extension…"	Dial-out [Sub dial-out]	1.0
`_do_extplay.wav`	"…please."	Dial-out [Sub dial-out]	0.8
`do_local_ringback.wav`	<sound of telephone ringing >	Dial-out (IP system only)	
`do_long_cmd_help.wav`	"After the call is answered, to connect the line into the conference, press star 1. To connect the line and continue dialing, press star 2. To disconnect the line, press star 3. To disconnect the line and continue dialing, press star 4."	Dial-out [Sub dial-out]	
`do_not_completed.wav`	"I'm sorry, your call could not be completed. Please try your call again."	Dial-out [Sub dial-out]	4.2
`do_not_enabled.wav`	"I'm sorry, this feature is not enabled."	Dial-out not enabled [Sub dial-out]	2.5
`do_num_verify_prompt_.wav`	"The number you dialed is…"	Dial-out [Sub dial-out]	1.5
`_do_num_verify_prompt.wav`	"Press pound to proceed, or press star to change this number."	Dial-out [Sub dial-out]	4.0
`do_number_prompt.wav`	"To dial out, please dial the area code and number that you wish to connect to the conference, followed by the pound sign. To return to the conference, press star."	Dial-out [Sub dial-out]	9.3

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`do_part_disconnected.wav`	"The participant was disconnected."	Dial-out (shortened call flow) [Sub dial-out short]	
`do_part_joined.wav`	"The participant has entered the conference."	Dial-out (shortened call flow) [Sub dial-out short]	
`do_proceed_prompt.wav`	"Press pound to proceed with dialing."	Dial-out [Sub dial-out]	2.8
`do_short_cmd_help.wav`	"Once the call is answered, to place the participant into the conference, press star 1. To disconnect the participant, press star 2."	Dial-out (shortened call flow) [Sub dial-out short]	
`do_short_number_prompt.wav`	"To enter the conference press star 1. To dial out to another participant, dial the number followed by the pound sign."	Dial-out (shortened call flow) [Sub dial-out short]	
`eea_anon_join.wav`	"An anonymous participant has joined the conference."	Name Announce [Entry noise]	3.3
`eea_anon_leave.wav`	"An anonymous participant has left the conference."	Name Announce [Exit noise]	3.3
`eea_change_failure.wav`	"I'm sorry, the system is currently unable to change this option, please try again later."	Entry/Exit Announcement changeable [Announce setting]	6.4
`eea_change_in_conf_not_enabled.wav`	"I'm sorry, this feature is not enabled."	Entry/Exit Announcement not changeable [Sub in conf]	2.5
`eea_change_not_enabled.wav`	"I'm sorry, this feature is not enabled."	Roll Call off [Roll call setting]	2.5
`eea_change_prompt.wav`	"To change the conference entry and exit announcement setting, press 2."	Entry/Exit Announcement changeable [Roll call setting]	3.9
`eea_name_entry.wav`	"Participants joining or leaving the conference are announced by name."	Entry/Exit Announcement changeable [Announce setting]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
eea_name_entry_no_rc.wav	"I'm sorry, Participant Name Record must be on to choose this option."	Entry/Exit Announcement changeable [Announce setting]	6.1
eea_name_join_.wav	<silence>	Name Announce [Entry announce]	
_eea_name_join.wav	"…has joined the conference."	Name Announce [Entry announce]	
eea_name_leave_.wav	<silence>	Name Announce [Exit announce]	
_eea_name_leave.wav	"…has left the conference."	Name Announce [Exit announce]	
eea_overview.wav	"The conference entry and exit announcement setting determines what will be heard when participants join and leave the conference. The choices are tones, name announce or silence."	Entry/Exit Announcement changeable [Account Options help]	13.8
eea_set_menu.wav	"To select name announce, press 1. To select tones, press 2. To select silence, press 3."	Entry/Exit Announcement changeable [Announce setting]	8.2
eea_silent_entry.wav	"Participants joining or leaving the conference are not announced."	Entry/exit announcement changeable [Announce setting]	
eea_tone_entry.wav	"Participants joining or leaving the conference are announced by tones."	Entry/Exit Announcement changeable [Announce setting]	
eea_tone_join.wav	<beeps>	Tone Announce	0.1
eea_tone_leave.wav	<beeps>	Tone Announce	0.1
error.wav	"error"	All	0.8
global_muted.wav	"The conference has been muted."	Custom mute/unmute confirmation (see page 162) [Sub in conf]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`global_unmuted.wav`	“The conference has been un-muted.”	Custom mute/unmute confirmation (see page 162) [Sub in conf]	
`helo_inbound.wav`	“Hello and welcome to the Conferencing Center.”	Private number initial greeting on a fixed access system. [Trad. Private, 2-Passwd. Private]	3.2
`helo_outbound.wav`	“Hello and welcome to the Conferencing Center.”	Dial-out	3.2
`hundred.wav`	“hundred”	All	0.7
`lck_conf_locked.wav`	“The conference has been locked.”	Conference Lock [Sub in conf]	1.2
`lck_conf_unlocked.wav`	“The conference has been unlocked.”	Conference Lock [Sub in conf]	1.8
`lck_locked_disconnect.wav`	“I’m sorry, the conference you are attempting to join has been locked. You have been disconnected by the system. Please try again.”	Conference Lock [Part call flow]	
`lo_entry_off.wav`	“Listen Only on Entry is off.”	Listen Only Entry [LO entry setting]	
`lo_entry_on.wav`	“Listen Only on Entry is on.”	Listen Only Entry [LO entry setting]	
`lo_entry_overview.wav`	“Listen Only on Entry places participants into conference in Listen Only mode.”	Listen Only Entry [Account Options help]	
`lo_entry_set_off_prompt.wav`	“To turn Listen Only on entry off, press 1.”	Listen Only Entry [LO entry setting]	
`lo_entry_set_on_prompt.wav`	“To turn Listen Only on entry on, press 1.”	Listen Only Entry [LO entry setting]	
`lo_part_join.wav`	“You are being placed into conference in listen only mode.”	Listen Only Entry [Participant call flow]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
lo_subs_join.wav	"This is a Listen Only conference."	Listen Only Entry [Sub call flow]	
mute_not_enabled.wav	"I'm sorry, this feature is not enabled."	Mute unavailable [In conf part or sub]	2.5
mute_off.wav	<beeps>	All [In conf part or sub]	
mute_on.wav	<beeps>	All [In conf part or sub]	
mute_part_join.wav	"You are being placed into conference in muted mode."	Mute All on [Participant call flow]	
numbers.wav	Indexed WAVE file containing all recorded numbers. See "Creating Your Own Indexed WAVE Files" on page 171.	All	
op_conf_doorbell.wav	"An operator wants to enter your conference. Please press star 5 to unlock it."	Operator available [In conf sub]	8.0
op_conf_leave.wav	<beep>	Played to operator when leaving conf	0.2
op_disconnect.wav	"You have been disconnected by the operator. Good-bye."	Operator available [Oper wait]	4.0
op_disconnect_from_chan.wav	<beep>	Played to operator when disconnecting from channel	0.2
op_gui_req.wav	"Your request will be answered by the next available operator. To cancel your request, press star zero."	Operator request via Moderator [In conf sub]	7.4
op_gui_req_cancel.wav	"Your operator request has been canceled."	Operator request canceled via Moderator [In conf sub]	2.3
op_hold.wav	"Please stand by for an operator."	Operator available [Oper wait]	2.3
op_listen.wav	<beep>	Played to operator when connecting to channel or conf	0.2

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
op_music.wav	<hold music>	Operator available [Oper wait]	30.0
op_no_op_disconnect.wav	"You have been disconnected by the system. Please try your call again."	All (no oper avail) [Oper wait]	4.4
op_not_available.wav	"I'm sorry, there are no operators available to service your request."	Operator request [In conf oper req]	4.2
op_req.wav	"Your request will be answered by the next available operator. To cancel your request, press star zero."	Operator request [In conf oper req]	7.4
op_req_cancel.wav	"Your operator request has been canceled."	Operator request [In conf oper req]	2.3
op_req_for_subs_only.wav	"I'm sorry, this feature is not enabled."	Operator request unavailable to part [In conf oper req]	2.5
op_req_not_enabled.wav	"I'm sorry, this feature is not enabled."	Operator request unavailable [In conf oper req]	2.5
pound.wav	"Pound"	All	0.4
press_#.wav	"Press pound."	All	
press_0.wav	"Press zero."	All	
press_9.wav	"Press 9."	All	
press_anykey.wav	"Press any key."	All	
press_star.wav	"Press star."	All	
private_muted.wav	"Your line is muted."	Custom mute/unmute confirmation (see page 162) [In conf part or sub]	
private_unmuted.wav	"Your line has been unmuted."	Custom mute/unmute confirmation (see page 162) [In conf part or sub]	
pt_operator.wav	"An operator…"	Roll Call (private) [In conf part or sub]	1.2

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`pt_recorder.wav`	"A recorder…"	Roll Call (private) [In conf part or sub]	2.0
`pw_confirmation.wav`	"Thank you."	Two-password private call flow [2-passwd private]	
`pw_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	Two-password private call flow [2-passwd private]	4.4
`pw_prompt.wav`	"Thank you for calling the conference center. Please enter your passcode followed by the pound sign at any time during this message."	Two-password private call flow [2-passwd private]	
`pw_wrong.wav`	"I'm sorry, that entry is not valid."	Two-password private call flow [2-passwd private]	3.0
`pw_wrong_.wav`	"I'm sorry, your entry…"	Two-password private call flow [2-passwd private]	1.9
`_pw_wrong.wav`	"… is not valid. Please enter the valid digits followed by the pound sign."	Two-password private call flow [2-passwd private]	5.2
`qs_change_prompt_.wav`	"To change the QuickStart setting…"	Quick Start [Account Options help]	2.6
`_qs_change_prompt.wav`	<silence>	Quick Start [Account Options menu]	
`qs_future_affecting.wav`	"…for future conferences. Current conferences are not affected by this change."	Quick Start [Quick Start setting]	5.9
`qs_off.wav`	"Quick Start is off."	Quick Start [Quick Start setting]	
`qs_on.wav`	"Quick Start is on."	Quick Start [Quick Start setting]	
`qs_overview.wav`	"Quick Start allows conferences to start as soon as the first participant dials in. Participants don't wait on hold for you to arrive."	Quick Start [Account Options help]	6.4

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	**Default Prompt**	**Configuration or Feature**	**Length (sec.)**
`qs_part_join.wav`	"This is a Quick Start conference, you will be placed into the conference."	Quick Start [Part call flow]	4.7
`qs_set_off_prompt.wav`	"To turn Quick Start off, press 1."	Quick Start [Quick Start setting]	
`qs_set_on_prompt.wav`	"To turn Quick Start on, press 1."	Quick Start [Quick Start setting]	
`qs_subs_join.wav`	"This is a Quick Start conference, you will be placed into the conference."	Quick Start [Sub call flow]	4.7
`rc_.wav`	"The following…"	Roll Call (private) [In conf part or sub]	
`_rc.wav`	"…participants are in conference."	Roll Call (private) [In conf part or sub]	
`rc_anon_count_.wav`	"The number of anonymous participants is…"	Roll Call (private) [In conf part or sub]	2.0
`rc_anonymous.wav`	"An anonymous participant."	Name Announce (Mod or oper clicks Play Name) [In conf sub]	2.0
`rc_change_prompt_.wav`	"To change Roll Call settings…"	Roll Call [Account Options menu]	1.8
`_rc_change_prompt.wav`	<silence>	Roll Call [Account Options menu]	
`rc_conf_.wav`	"The following…"	Roll Call (conf) [In conf sub]	
`_rc_conf.wav`	"…participants are in conference."	Roll Call (conf) [In conf sub]	
`rc_conf_anon_count_.wav`	"The number of anonymous participants is…"	Roll Call (conf) [In conf sub]	2.0
`rc_identify_prompt.wav`	"After the tone, please state your name, followed by the pound sign. <bong>"	Roll Call [Name Record]	2.0
`rc_off.wav`	"Roll Call is off."	Roll Call [Roll Call setting]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`rc_on.wav`	"Roll call is on. Participants will be prompted to record their names."	Roll Call [Roll Call setting]	
`rc_overview.wav`	“Roll Call prompts participants to record their names as they join a conference. During the conference, any participant can press star 9 to hear the roll call replayed privately.”	Roll Call [Account Options help]	
`rc_rerecord_prompt.wav`	“I’m sorry, we did not get your name.”	Roll Call [Name Record]	2.0
`rc_set_off_prompt.wav`	“To turn Roll Call off, press 1.”	Roll Call [Roll Call setting]	
`rc_set_on_prompt.wav`	"To turn roll call on so participants are prompted to record their names, press 1."	Roll Call [Roll Call setting]	
`rec_change_failed.wav`	“I’m sorry, your call could not be completed. Please try your call again.”	Recorder Dial Out [Recording]	
`rec_conf_full.wav`	“I’m sorry. Your conference is full. A recording connection cannot be added.”	Recorder Dial Out [Recording]	5.7
`rec_not_enabled.wav`	“I’m sorry, this feature is not enabled.”	Recorder Dial Out unavailable [Recording]	2.5
`rec_part_join_reminder.wav`	“This conference is being recorded.”	Recorder Dial Out [Part call flow]	2.3
`rec_rejoin_reminder.wav`	“This conference is being recorded.”	Recorder Dial Out [In conf rejoin]	2.3
`rec_start_prompt.wav`	“To start the conference recording, press 1. To cancel, press star.”	Recorder Dial Out [Recording]	5.7
`rec_started.wav`	“This conference is being recorded.”	Recorder Dial Out [Recording]	2.3
`rec_stop_prompt.wav`	“To stop the conference recording, press 1. To cancel, press star.”	Recorder Dial Out [Recording]	5.7
`rec_stopped.wav`	“This conference is no longer being recorded.”	Recorder Dial Out [Recording]	3.0
`rec_subs_join_reminder.wav`	“This conference is being recorded.”	Recorder Dial Out [Sub call flow]	2.3

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`rec_wait.wav`	"Please stand by while your recording connection is established. To cancel the recording, press star 2."	Recorder Dial Out [Recording]	7.2
`sd_chan_.wav`	"Due to technical difficulties, you will be disconnected in…"	All	
`_sd_chan.wav`	"…minutes."	All	
`sd_chan_now.wav`	"Due to technical difficulties, you will be disconnected now."	All	
`sd_conf_.wav`	"Due to technical difficulties, your conference will end in…"	All	
`_sd_conf.wav`	"…minutes."	All	
`sd_conf_now.wav`	"Due to technical difficulties, your conference will end now."	All	
`silence.wav`	<silence>	Not used as named. Copy and rename to replace any prompt you don't want played.	
`spw_change_failure.wav`	"The system is unable to change your password at this time. Please try again later."	Password changeable [PW setting]	5.0
`spw_change_prompt_.wav`	"To change your subscriber password…"	Password changeable [Account Options menu]	2.6
`_spw_change_prompt.wav`	<silence>	Password changeable [Account Options menu]	
`spw_disconnect.wav`	"You have been disconnected by the system. Please try your call again."	Prompt for Subscriber [Sub validation]	4.4
`spw_invalid.wav`	"I'm sorry, the password you entered is not permitted. Please enter a different password followed by the pound sign."	Password changeable [PW setting]	6.7

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`spw_overview.wav`	"The subscriber password is the password you use to log into the conferencing system, to start and join your conferences, and to change your account options."	Password changeable [Account Options help]	
`spw_prompt.wav`	"Please enter your subscriber password followed by the pound sign."	Prompt for Subscriber [Sub validation]	3.7
`spw_repeat_.wav`	"Your new subscriber password is…"	Password changeable [PW setting]	2.4
`spw_set_prompt.wav`	"Please enter your new subscriber password followed by the pound sign. The password must be 4 to 20 digits."	Password changeable [PW setting]	7.3
`spw_wrong.wav`	"I'm sorry, that entry is not valid."	Prompt for Subscriber [Sub validation]	3.0
`spw_wrong_.wav`	"I'm sorry, your entry…"	Prompt for Subscriber [Sub validation]	1.9
`_spw_wrong.wav`	"… is not valid. Please enter the valid digits followed by the pound sign."	Prompt for Subscriber [Sub validation]	5.2
`spw_wrong_length.wav`	"I'm sorry, the subscriber password must be between 4 and 20 digits long. Please re-enter your password followed by the pound sign."	Password changeable [PW setting]	8.6
`star.wav`	"Star."	All	0.5
`sys_error.wav`	"System error."	All	0.9
`wr_change_prompt_.wav`	"To change the conference lock type setting…"	Waiting Room [Account Options menu]	
`_wr_change_prompt.wav`	<silence>	Waiting Room [Account Options menu]	
`wr_cmd_help.wav`	"Press pound 5 to process callers in the waiting room. Press pound 6 to toggle notification announcements on or off."	Waiting Room [Sub call flow, Sub WR processing]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`wr_conf_locked.wav`	"This conference has been locked with Waiting Room on."	Waiting Room [In conf sub]	
`wr_count_.wav`	"There are…"	Waiting Room [Sub WR processing]	
`_wr_count.wav`	"…participants in the waiting room."	Waiting Room [Sub WR processing]	
`wr_empty.wav`	"There are no callers in the waiting room at this time. You will now be returned to the main conference."	Waiting Room [Sub WR processing]	
`wr_handling_prompt.wav`	"Callers in the waiting room are processed in the order they arrived."	Waiting Room [Sub WR processing]	
`wr_hold.wav`	"This conference is locked. Please hold while the Subscriber is notified."	Waiting Room [Part lock/full processing]	
`wr_music.wav`	<hold music>	Waiting Room [Part lock/full processing]	30.0
`wr_new_anon_caller.wav`	"An anonymous participant would like to join your call. Press pound 5 to process callers in the waiting room. Press pound 6 to toggle caller notification announcements on or off."	Waiting Room [Sub WR processing]	
`wr_new_caller_.wav`	"A new caller…"	Waiting Room [Sub WR processing]	
`_wr_new_caller.wav`	"…would like to join your conference. Press pound 5 to process callers in the waiting room. Press pound 6 to turn caller notification off."	Waiting Room [Sub WR processing]	
`wr_notification_off.wav`	"Waiting room notification is off."	Waiting Room [Sub WR processing]	
`wr_notification_on.wav`	"Waiting room notification is on."	Waiting Room [Sub WR processing]	

Table A-1 All voice prompts, alphabetized (continued)

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
wr_off.wav	"Currently, your conferences start unlocked, so participants are put directly into conference."	Lock (Waiting Room) available [WR setting]	
wr_on_entry.wav	"Currently, your conferences start locked, so participants are put into the waiting room."	Lock (Waiting Room) available [WR setting]	
wr_on_entry_conf_start.wav	"The conference will start locked with the waiting room on."	Waiting Room [Sub call flow]	
wr_on_lock.wav	"Currently, your conferences start unlocked, so participants are put directly into the conference, but the waiting room is available for later use."	Lock (Waiting Room) available [WR setting]	
wr_overview.wav	"The conference lock type option determines if callers dialing into a locked conference are disconnected or put into a waiting room."	Waiting Room [Account Options help]	
wr_private_conf_cmd_ help.wav	"Press pound to talk with the next caller. Then, press 1 to join the caller to conference. Press 2 to leave the caller in the Waiting Room. Press 3 to disconnect the caller. Press star to return to the main conference."	Waiting Room [Sub WR processing]	
wr_private_conf_w_op.wav	"You are being placed in a private conference with an operator. Please hold."	Waiting Room [Part lock/full processing]	
wr_private_conf_w_subs.wav	"You are being placed in a private conference with the Subscriber. Please hold."	Waiting Room [Part lock/full processing]	
wr_set_off_prompt.wav	"To start conferences unlocked with waiting room off, press 2. Participants will be put directly into conference."	Lock (Waiting Room) available [WR setting]	
wr_set_on_entry_prompt.wav	"To start conferences locked, so participants are put into the waiting room, press 1."	Lock (Waiting Room) available [WR setting]	
wr_set_on_lock_prompt.wav	"To start conferences unlocked with waiting room on, press 3. Participants will be put directly into conference, but the waiting room is available for later use."	Lock (Waiting Room) available [WR setting]	

Table A-1 *All voice prompts, alphabetized (continued)*

Required Filename	Default Prompt	Configuration or Feature	Length (sec.)
`wr_sub_hold.wav`	"The subscriber has returned you to the waiting room. Please hold."	Waiting Room [Part lock/full processing]	
`wr_subs_disconnect.wav`	"You have been disconnected by the conference moderator. Good bye."	Waiting Room [Part lock/full processing]	
`wr_timeout_disconnect.wav`	"Sorry. You have been in the waiting room for the maximum amount of time and will be disconnected. Goodbye."	Waiting Room [Part lock/full processing]	

Call Flow Diagrams

The following pages contain call flow diagrams for the ReadiVoice system. Figure A-2 provides a legend that explains some of the symbols and conventions used in the diagrams.

The diagrams themselves are arranged into several groups or categories, as listed in Table A-2 below.

Table A-2 *Guide to call flow diagrams*

Category	Figures	Description
Initial Inbound Call Flows	A-3 – A-8	Initial inbound call processing for different call flow configurations. During this processing, the system identifies the caller's conference and determines whether the caller is the subscriber or a participant.
Dial-out (Outbound) Call Flows	A-9 – A-13	Processes related to dial-out, including dial-out to a recording device.
Subscriber and Participant Call Flows	A-14 – A-22	Subscriber and participant call flows after the initial entry processing and some of the subprocesses within these call flows.
Subscriber Account Options Processes	A-23 – A-32	Processes related to the subscriber's account options menu and the settings available from it.
In-conference processes	A-33 – A-46	Processes related to in-conference actions by the subscriber, participants, or moderator, plus the system's conference termination process and other in-conference events.

Figure A-2 Understanding the call flow diagrams

Call Flow Legend

Examples of and information about certain symbols and conventions used in the call flow diagrams.

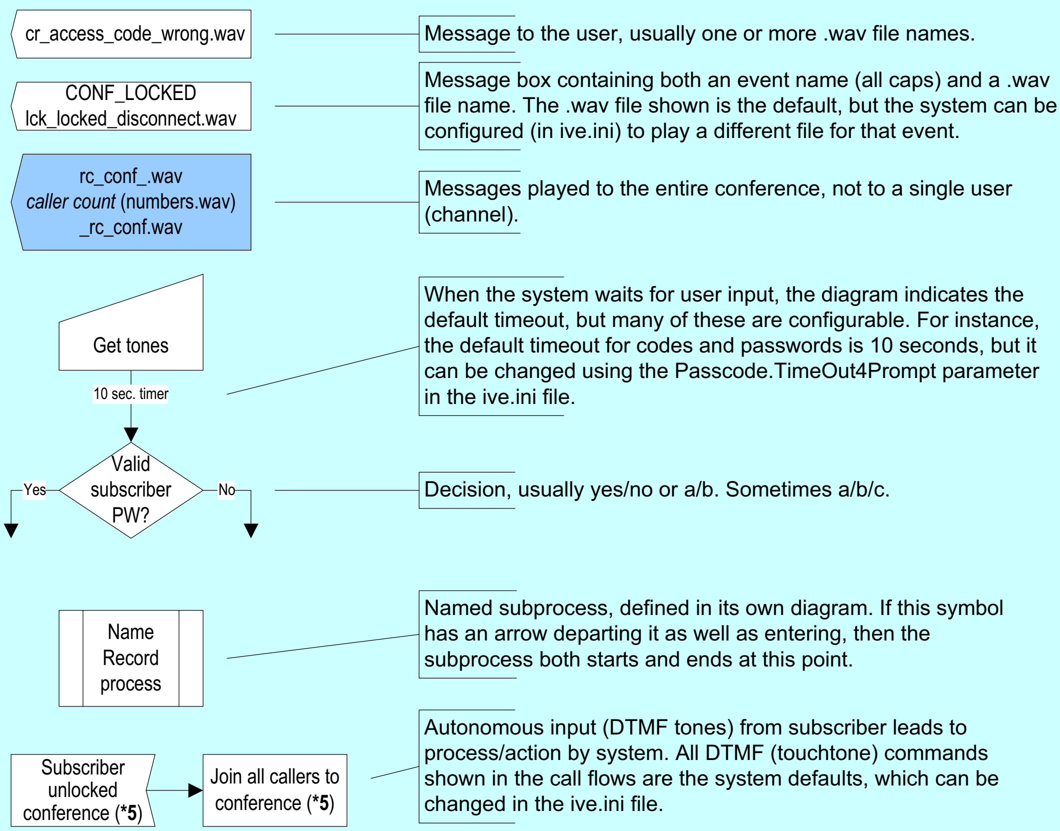
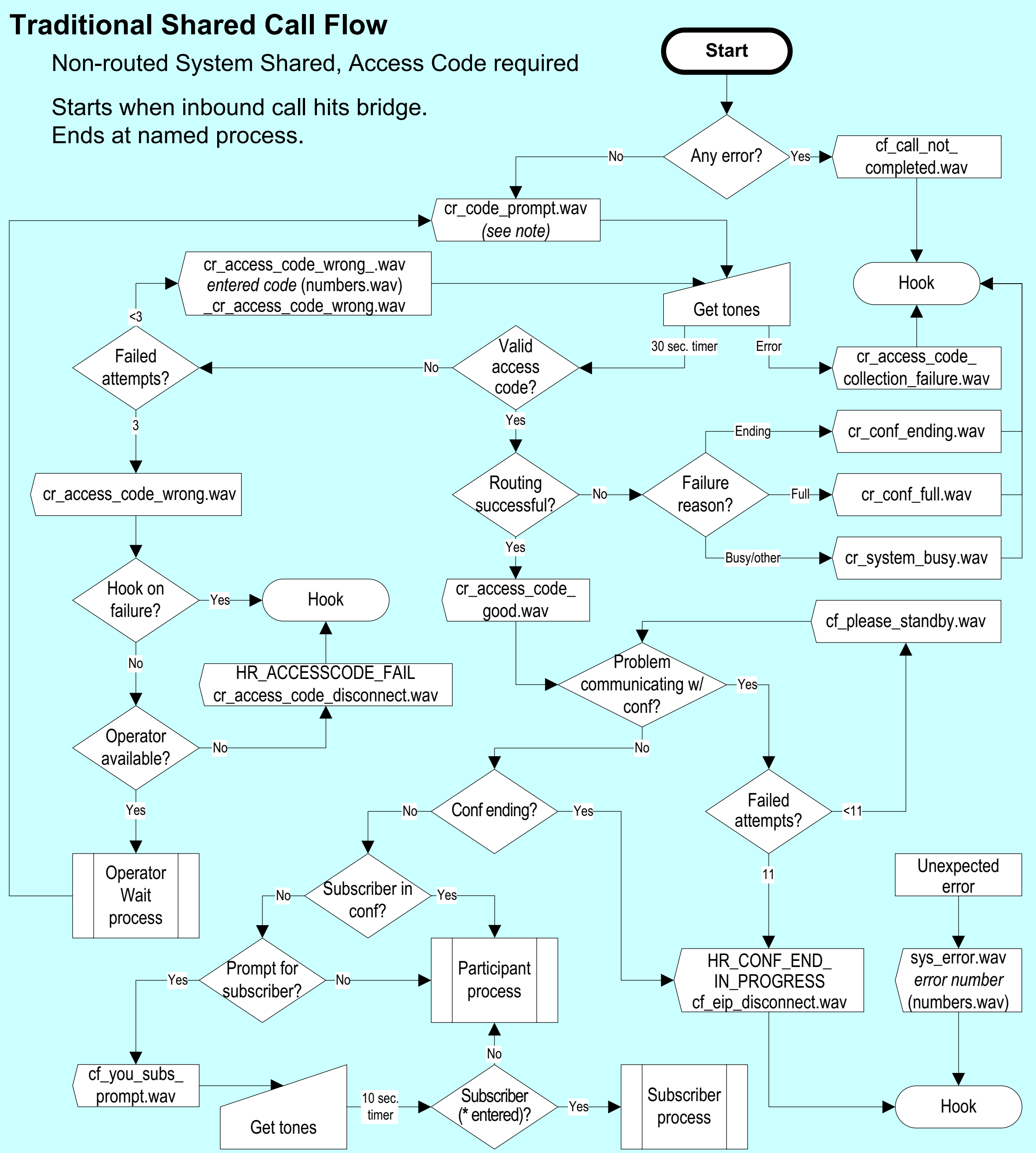


Figure A-3 *Traditional shared call flow (initial entry)*



Note: The initial greeting, cr_code_prompt.wav, is a symbolic link to either cr_access_code_prompt.wav or pw_prompt.wav, depending on system configuration. See "About Initial Greetings" on page 181 for details.

Figure A-4 *Traditional private call flow (initial entry)*

Traditional Private Call Flow

Non-Routed System Private, Participant Password not available

Starts when inbound call hits bridge.
Ends at named process.

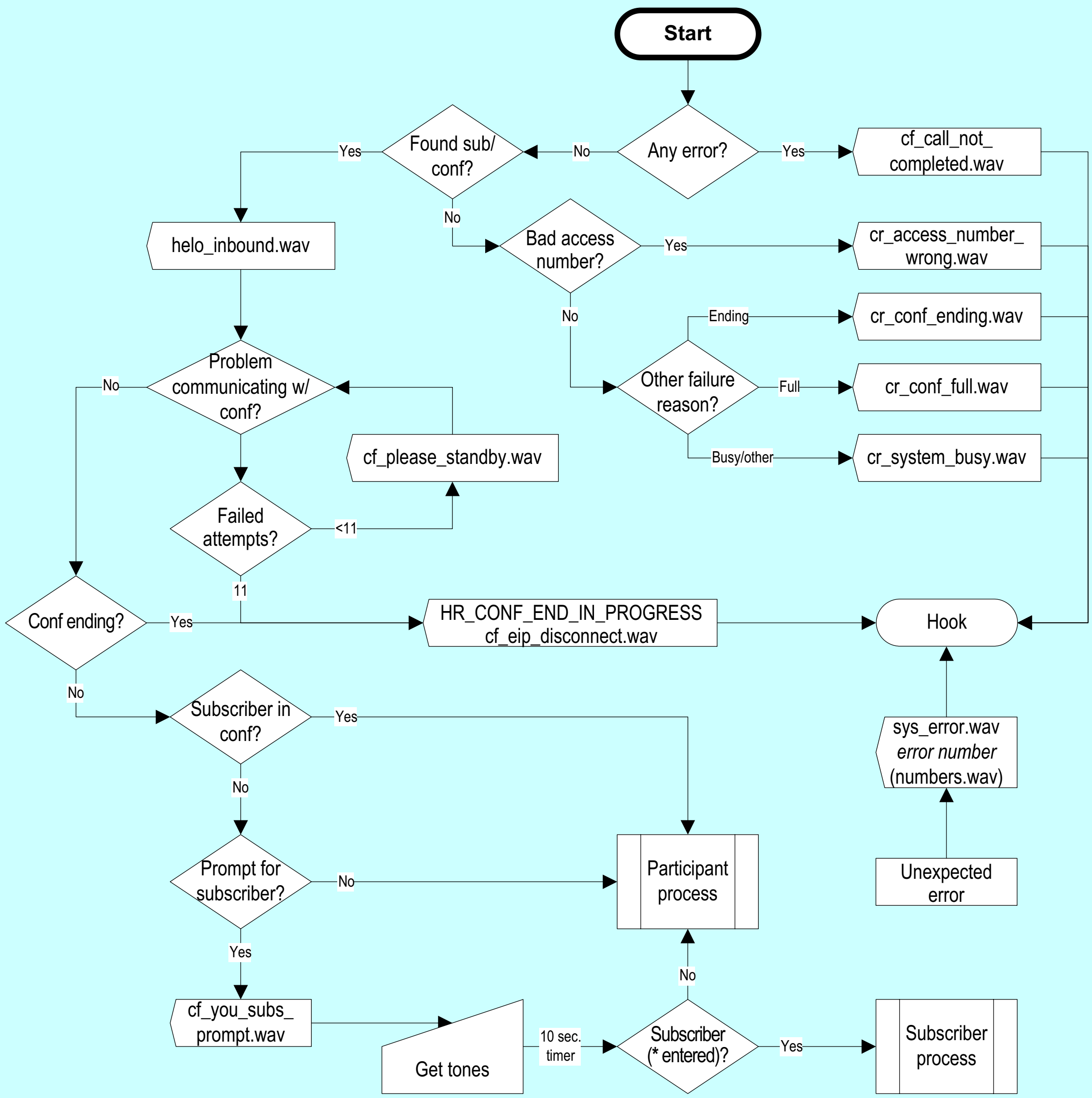
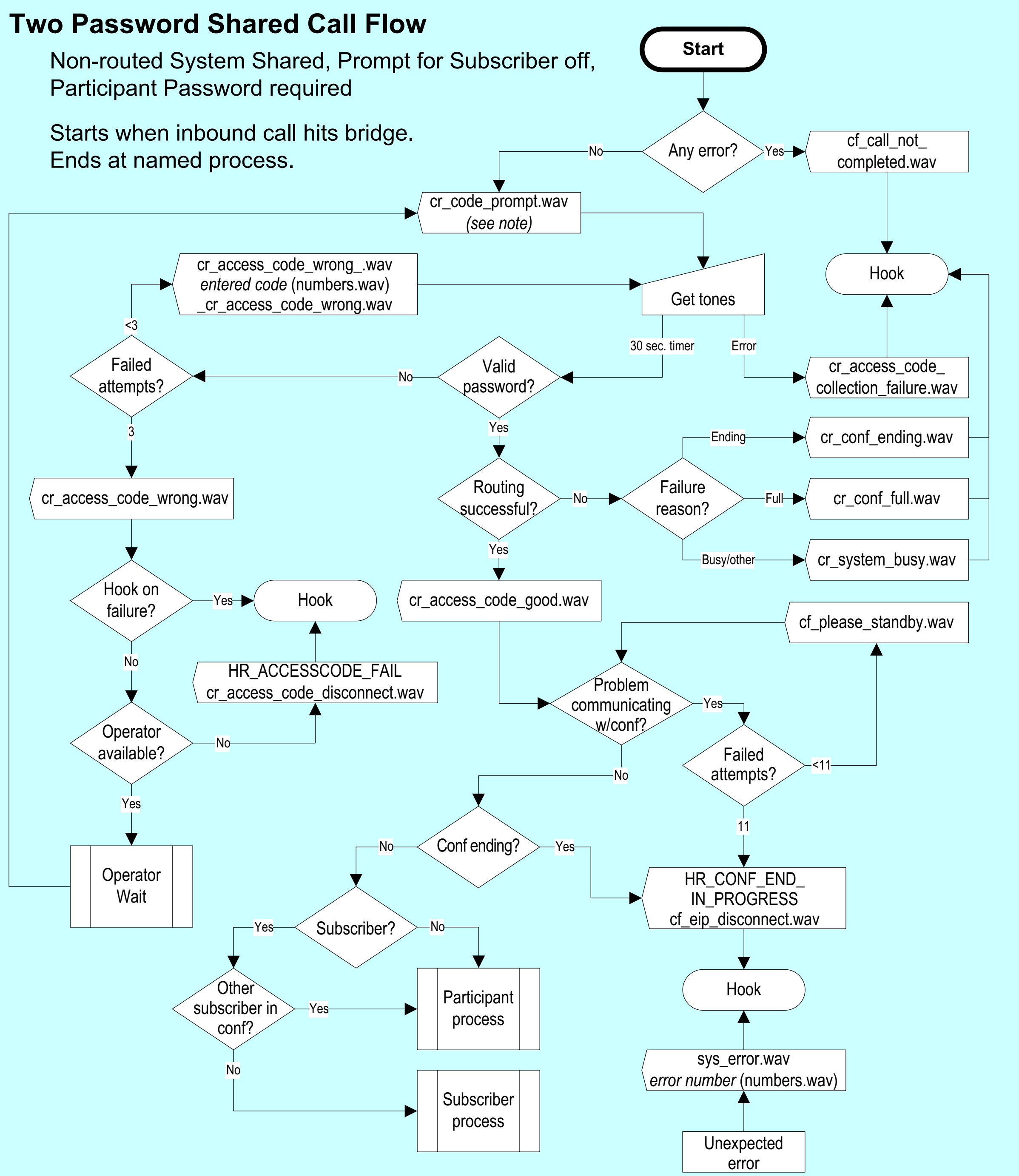


Figure A-5 *Two-password shared call flow (initial entry)*



Note: The initial greeting, cr_code_prompt.wav, is a symbolic link to either cr_access_code_prompt.wav or pw_prompt.wav, depending on system configuration. See "About Initial Greetings" on page 181 for details.

Figure A-6 *Two-password private call flow (initial entry)*

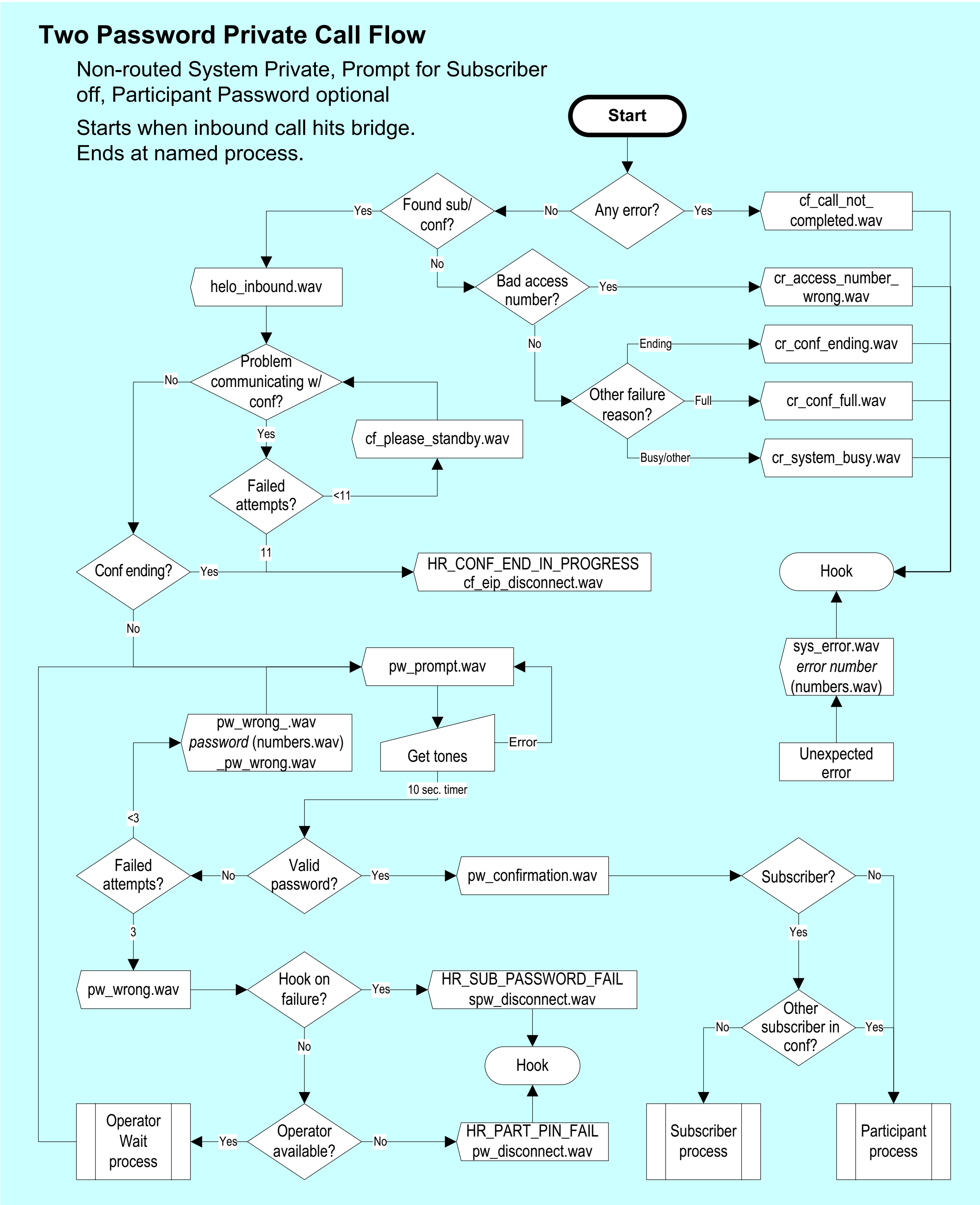


Figure A-7 *Routed (INCR) call flow (initial entry)*

Routed Call Flow

Routed (INCR) System – Access Code, if any, collected by switch.

Starts when inbound call hits bridge. Ends at named process.

Figure A-8 *IP Call process*

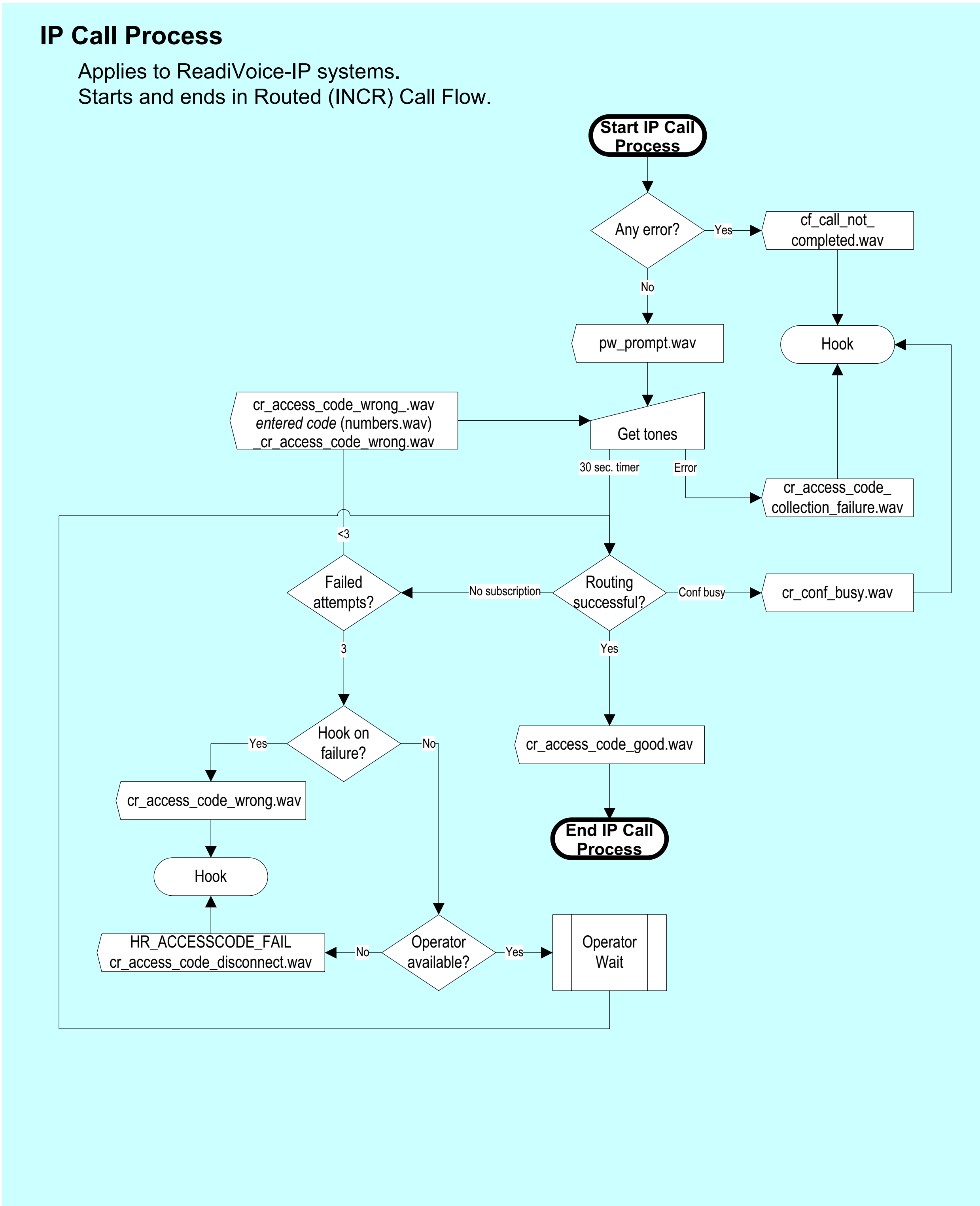


Figure A-9 *Recording dial-out process*

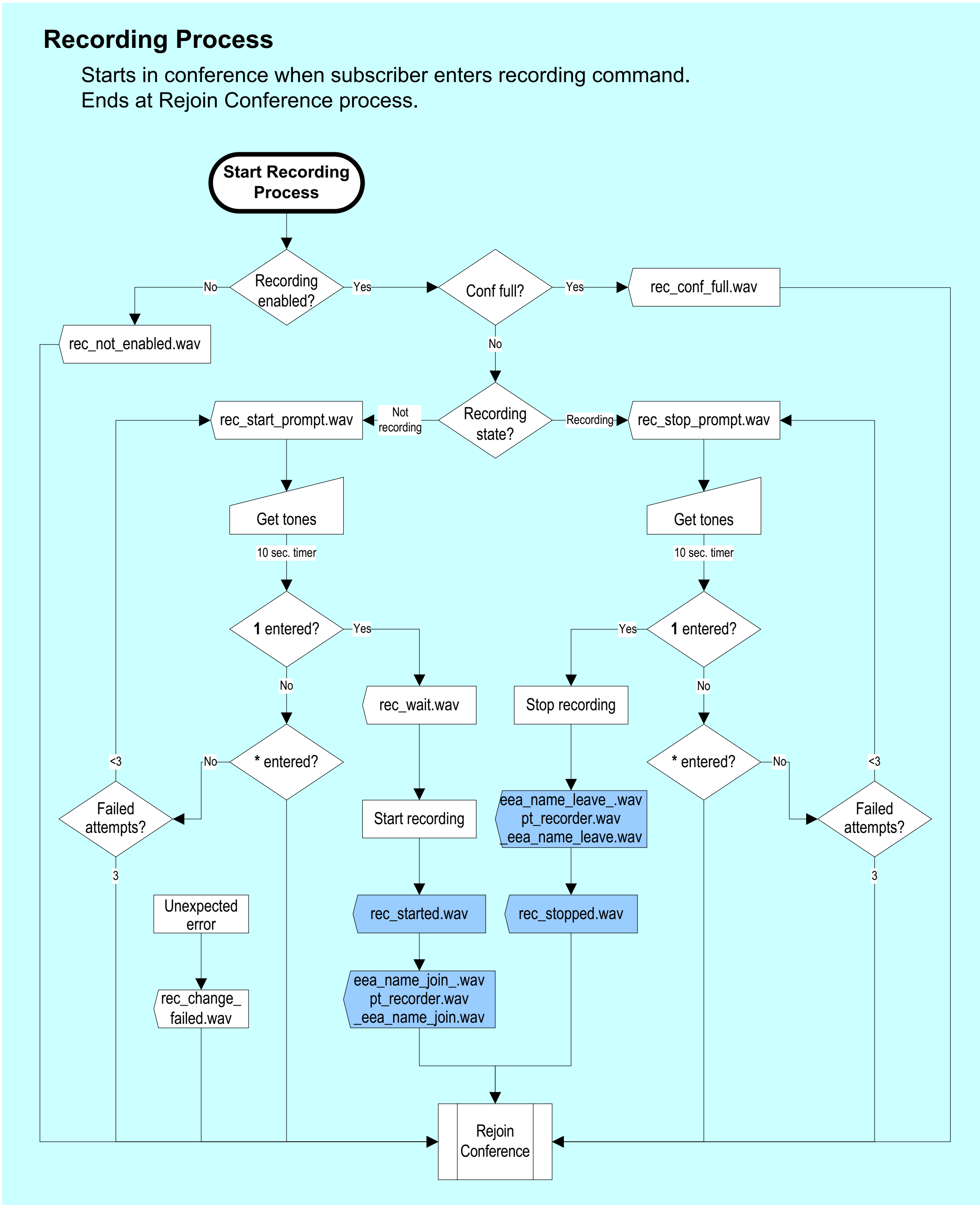


Figure A-10 *Subscriber dial-out process (long)*

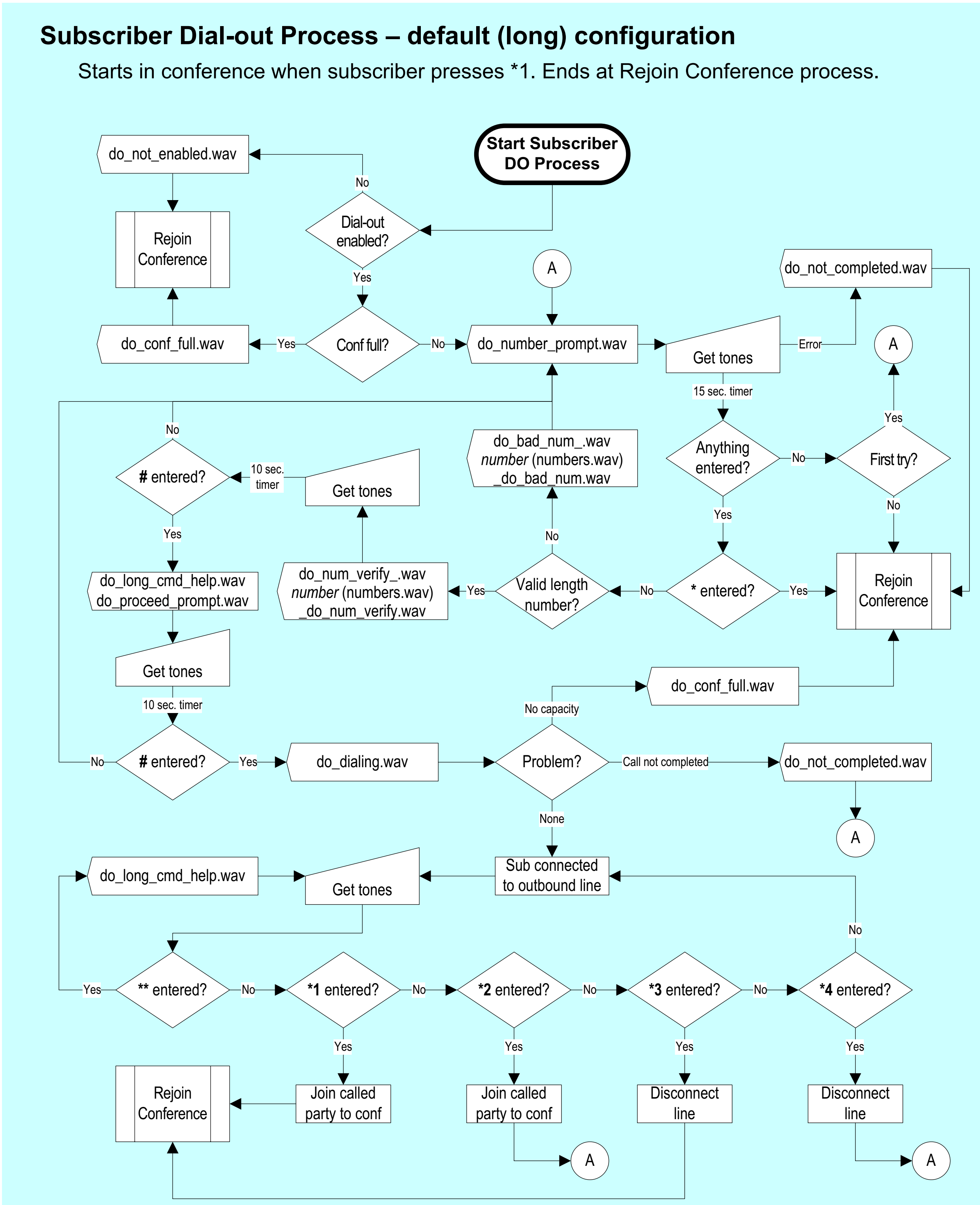


Figure A-11 *Subscriber dial-out process (short)*

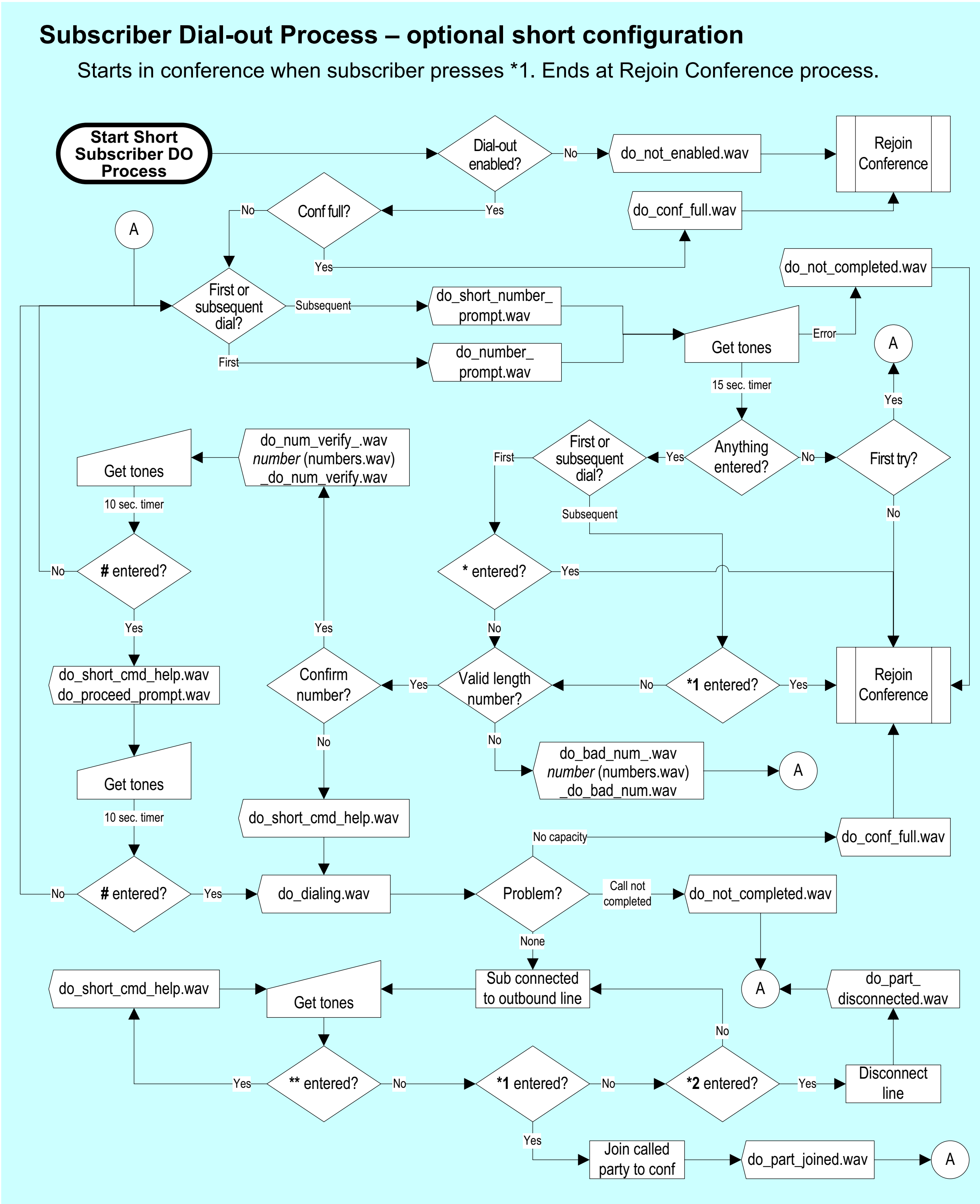


Figure A-12 *Called party dial-out process*

Called Party Dial-out Process

Describes system interaction with the dialed-out line (the called party). Starts when system accesses line to place outbound call. May be initiated via Operator or Moderator interface or via “One Click” link. Ends at named process.

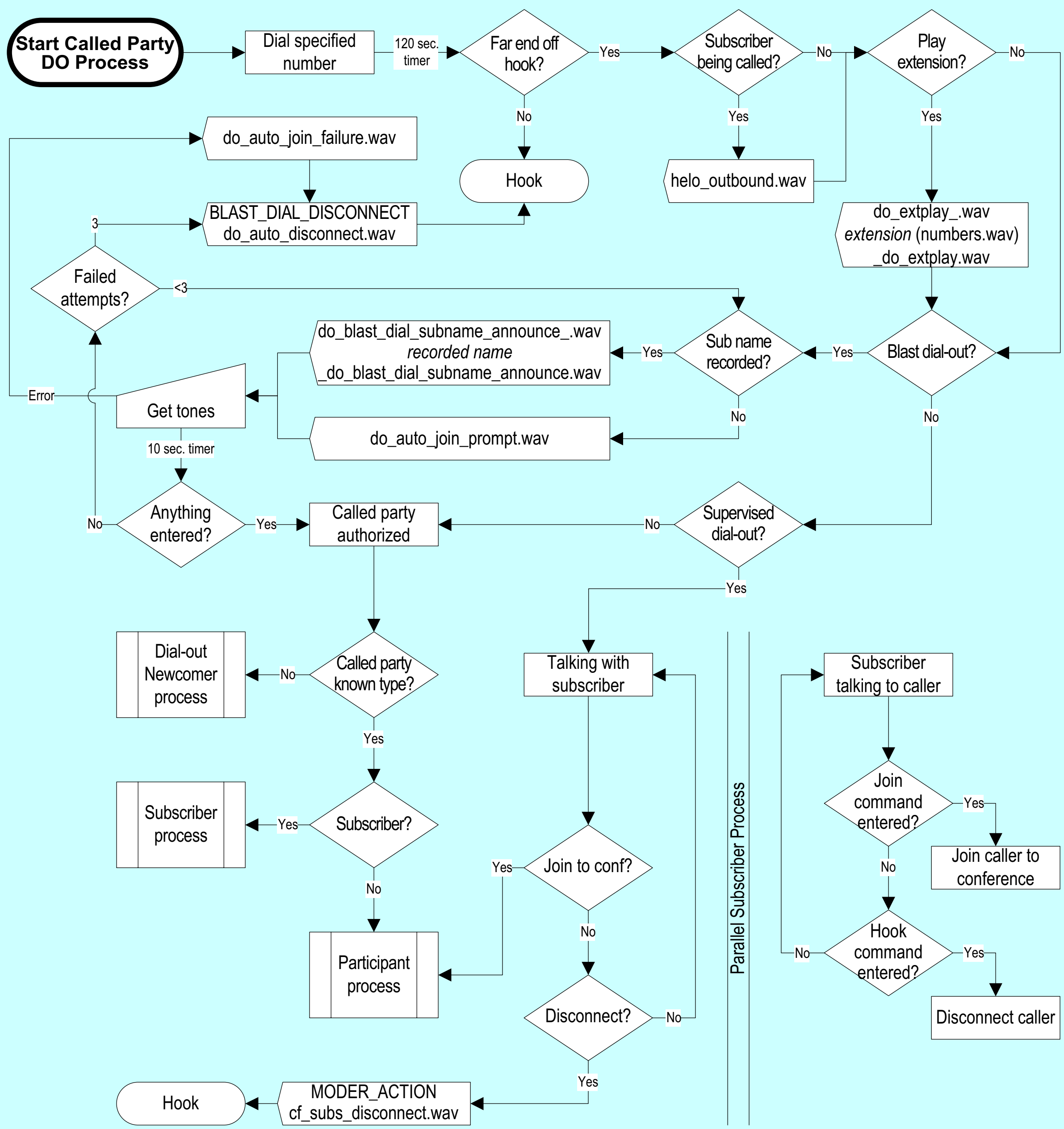


Figure A-13 *Dial-out newcomer process*

Dial-Out Newcomer Process

Starts in Called Party Dial-Out process when caller type (subscriber or participant) isn't known. Ends at named process.

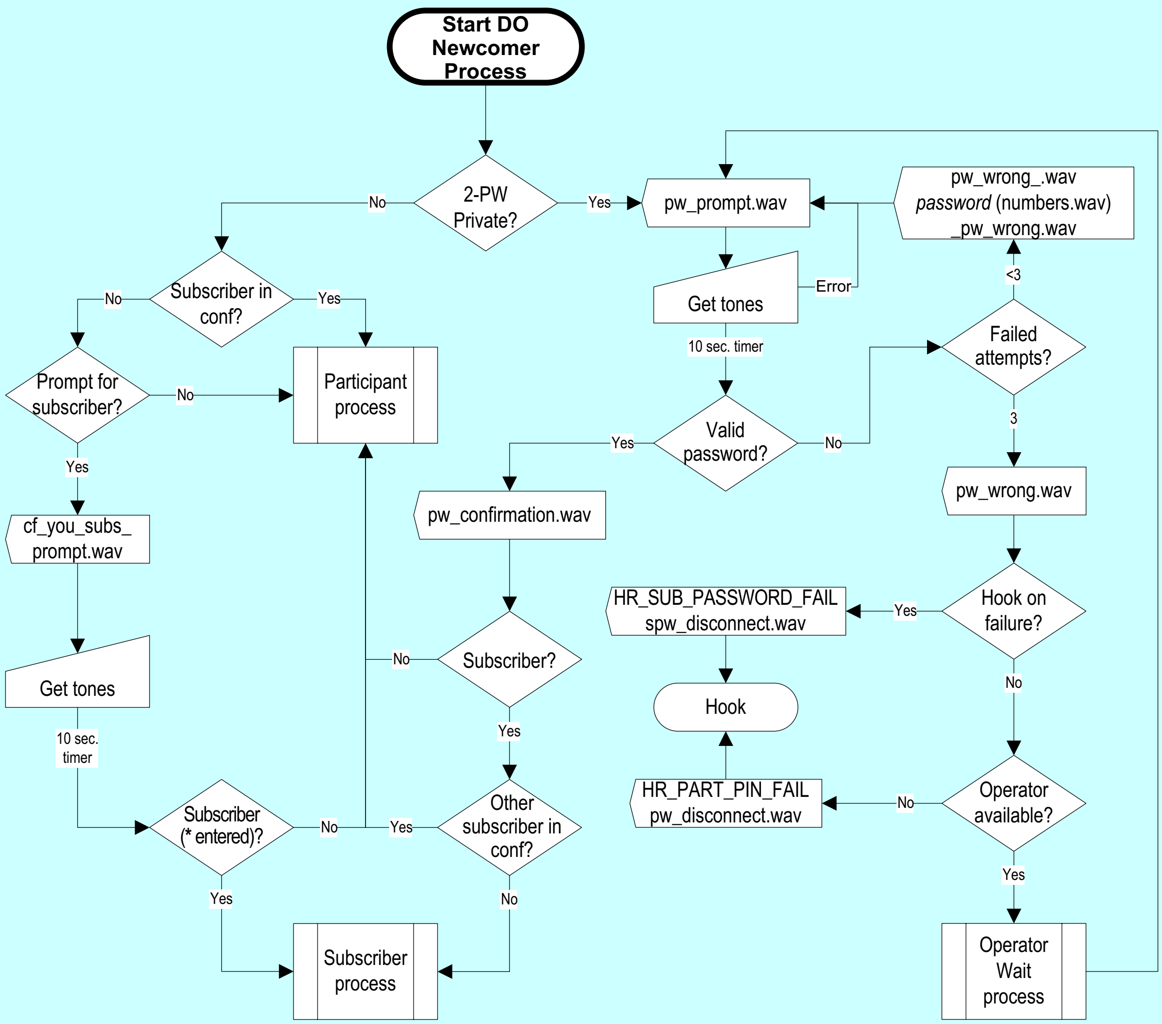


Figure A-14 *Subscriber call flow (after initial entry)*

Figure A-15 *Participant call flow (after initial entry)*

Participant Process

Starts after initial entry process (inbound call flow or called party dial-out process) or in Subscriber Validation process. Named processes are defined in their own flow diagrams.

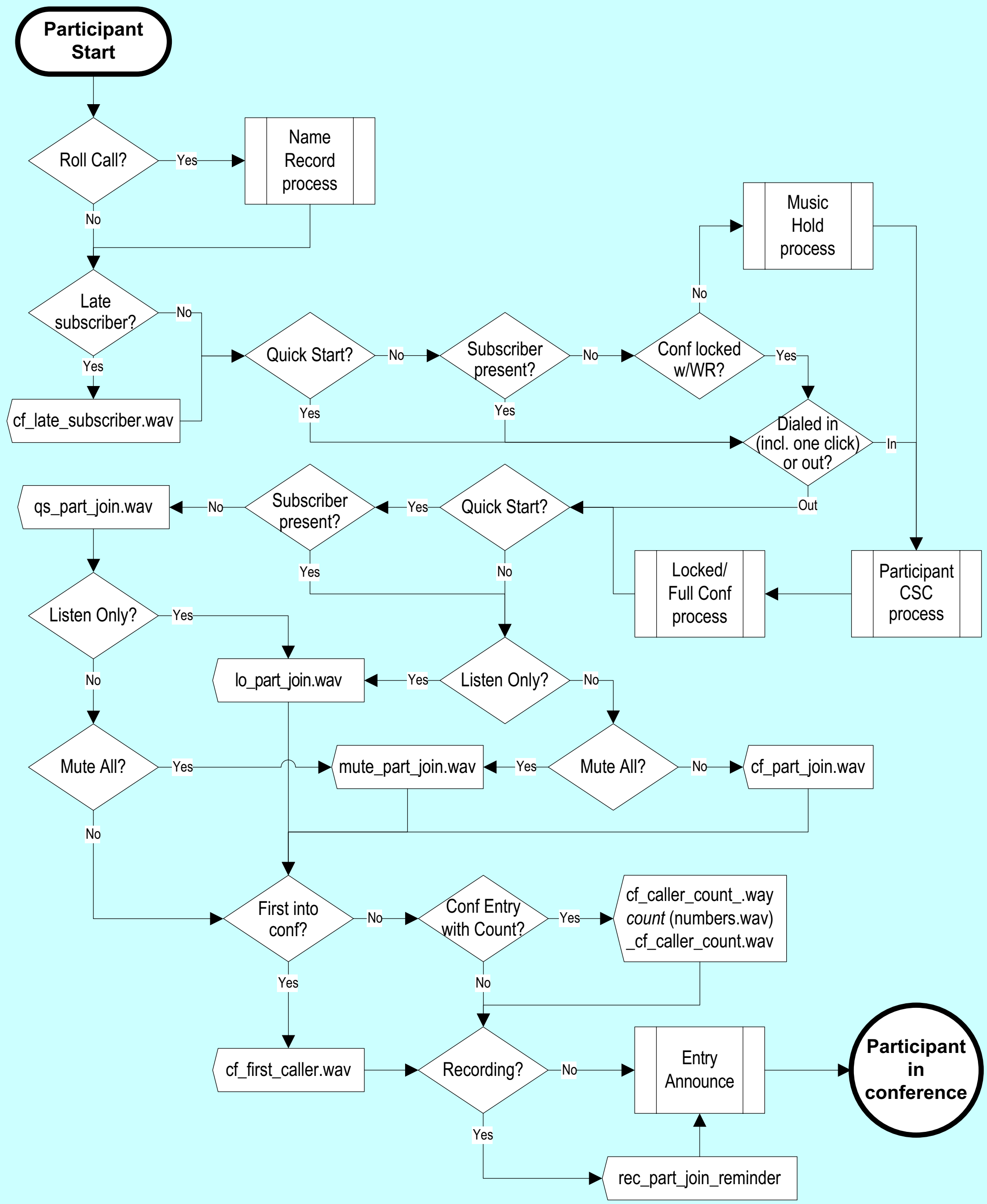


***Figure A-16** Subscriber validation process*

Subscriber Validation

Starts in Subscriber Call Flow or Music Hold.
Ends in Subscriber Call Flow.

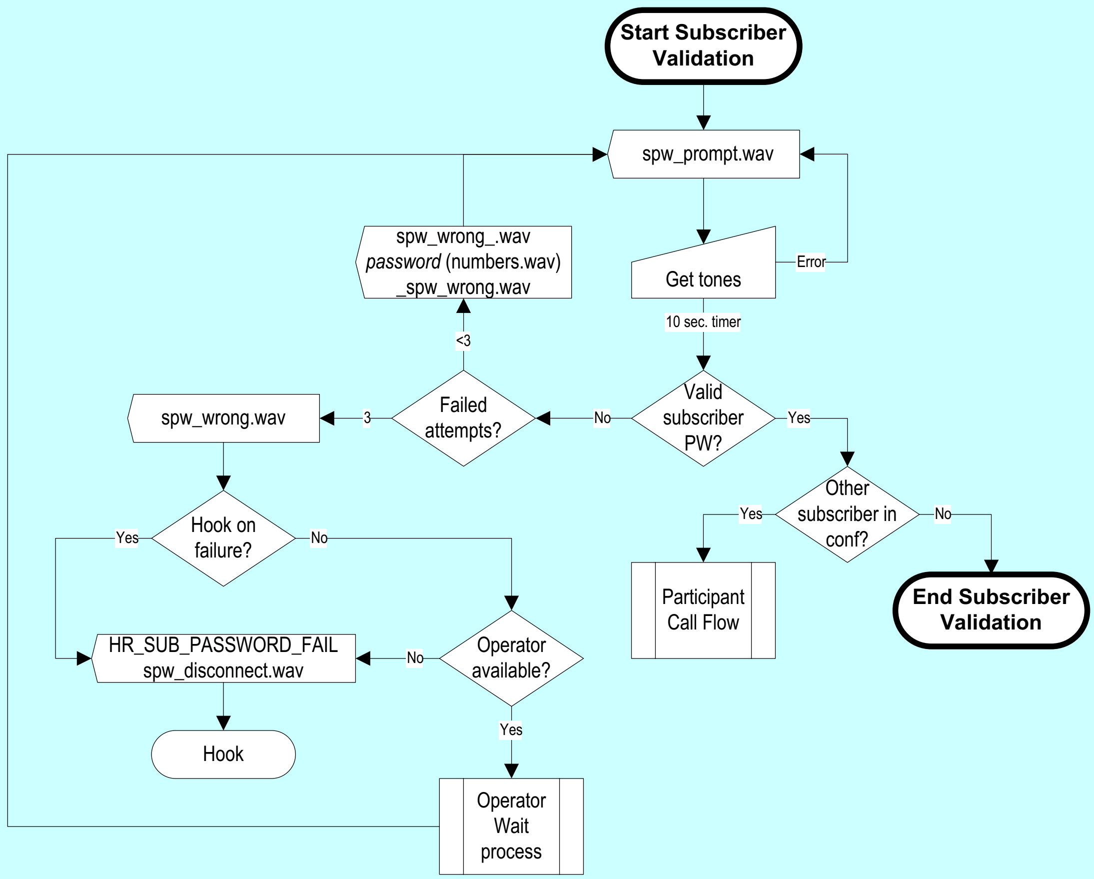


Figure A-17 *Conference security code process (subscriber)*

Figure A-18 *Conference security code process (participant)*

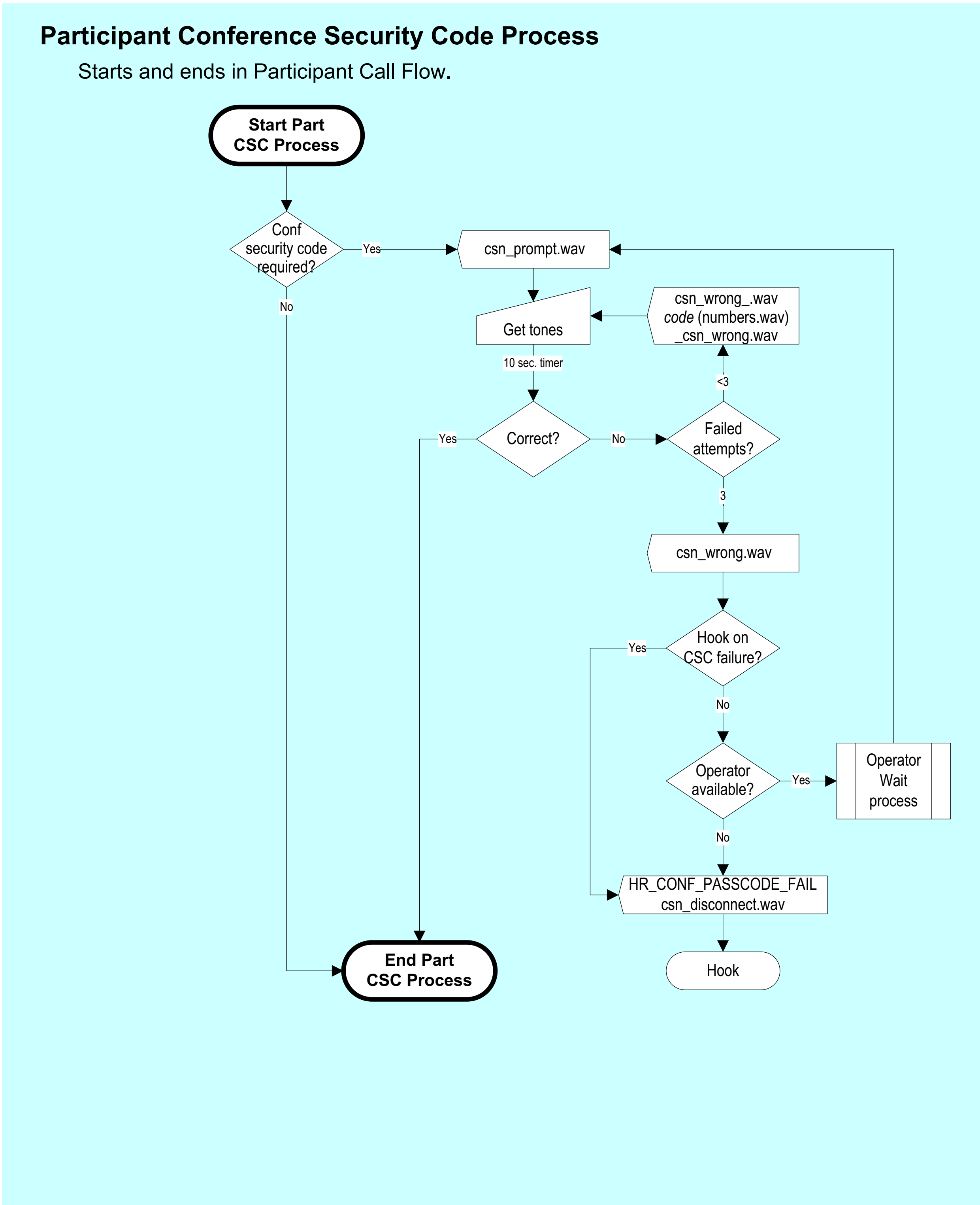


Figure A-19 *Music hold process*

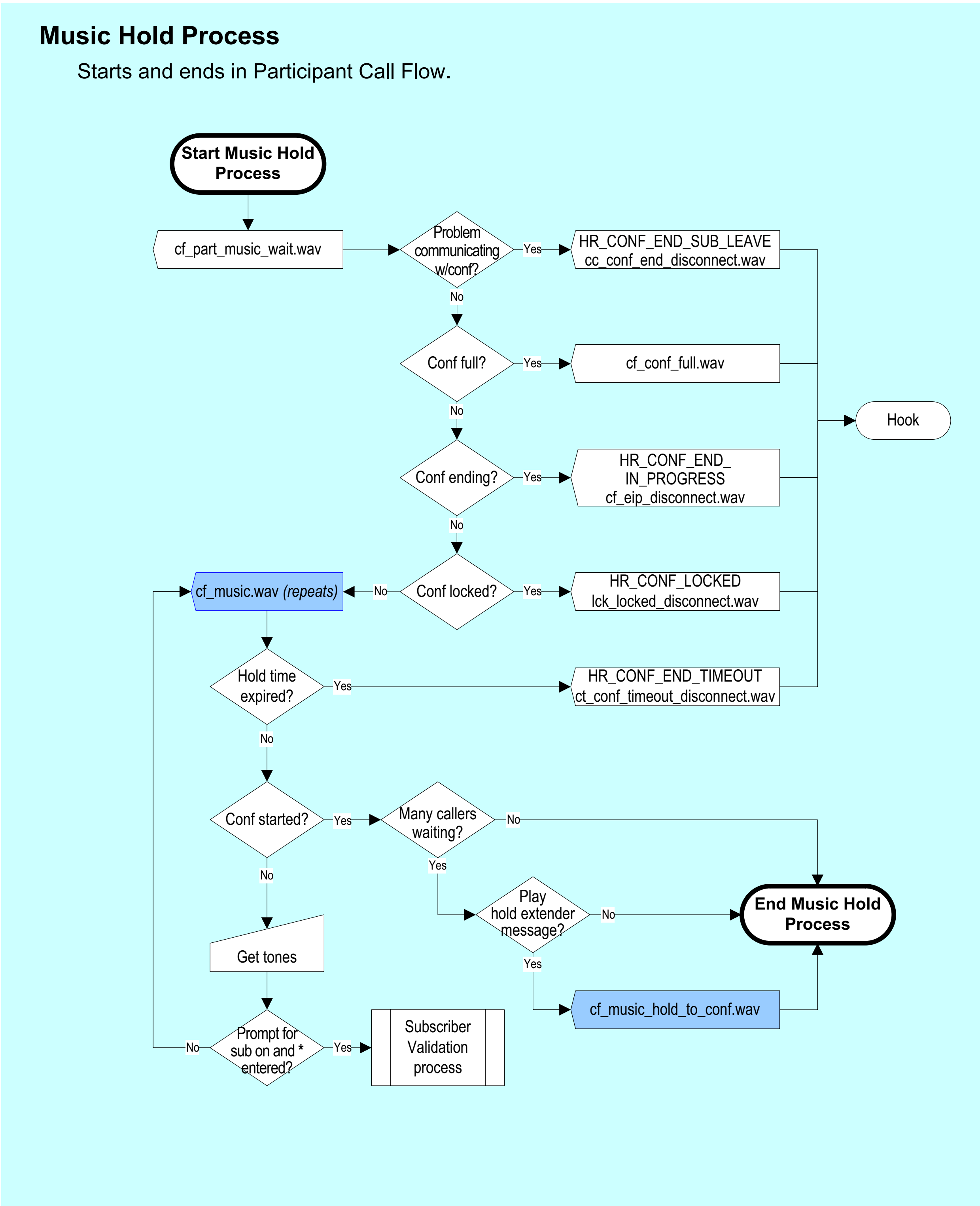


Figure A-20 Name record process

Subscriber/Participant Name Record Process

Starts and ends in Subscriber or Participant Call Flow.

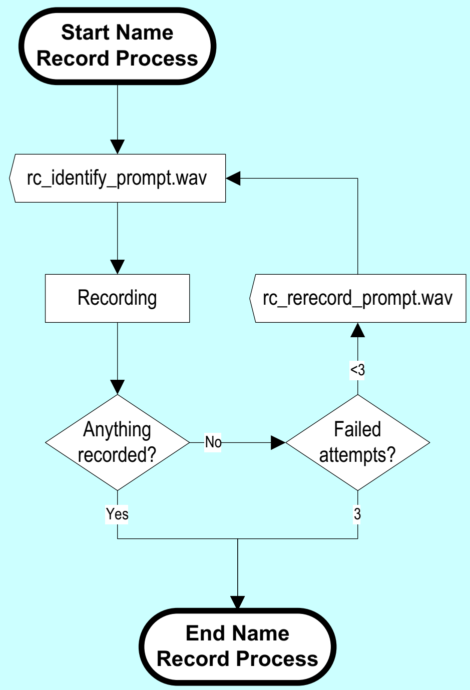


Figure A-21 *Locked/full conference process (participant)*

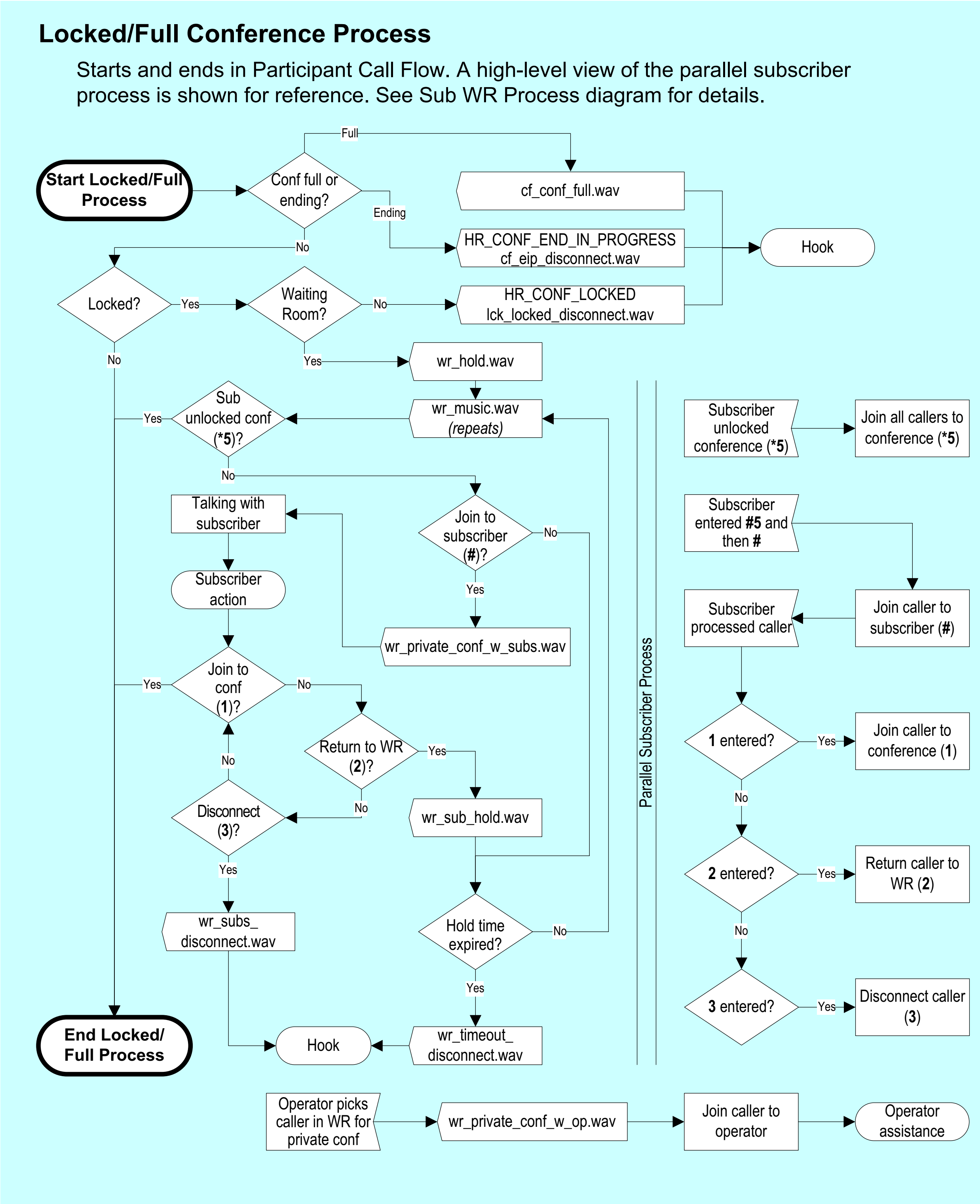


Figure A-22 Operator wait process

Operator Wait Process

Starts in Traditional Shared, Two Password Shared, Two Password Private, Routed, IP Call, Subscriber Validation, or Participant Conference Security Code.
Ends where it started.

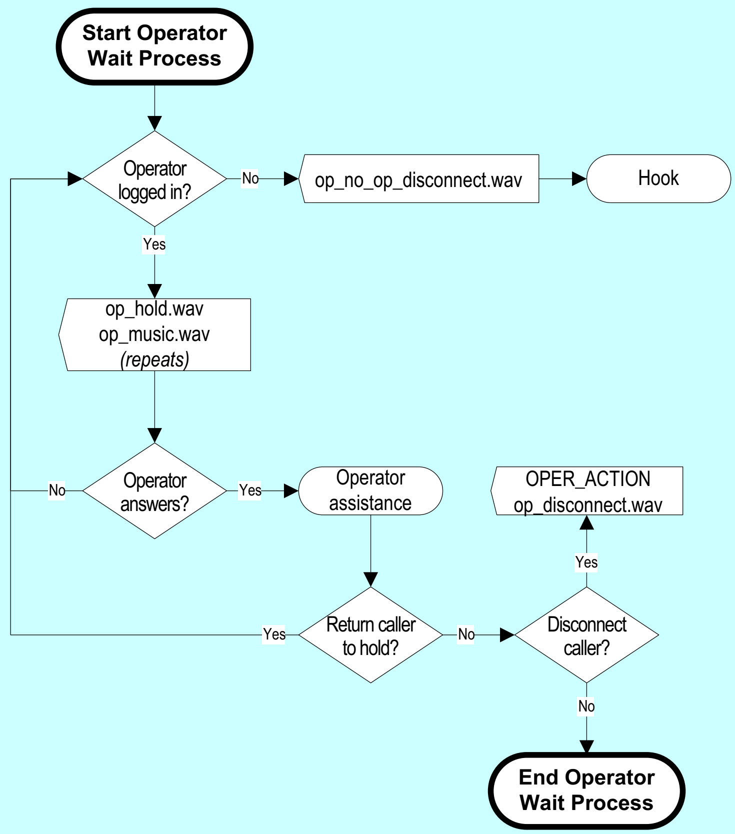


Figure A-23 Account options process

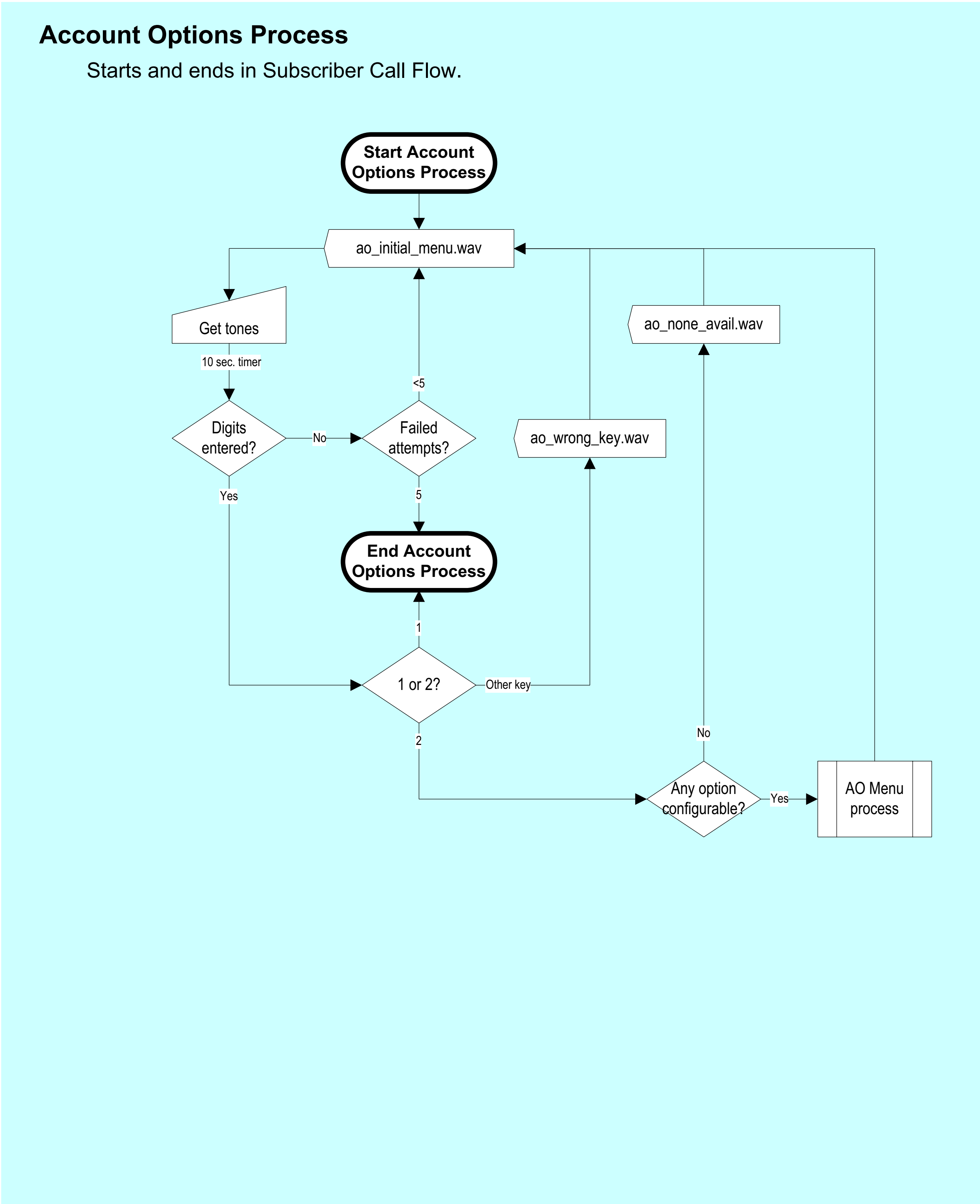


Figure A-24 *Account options menu*

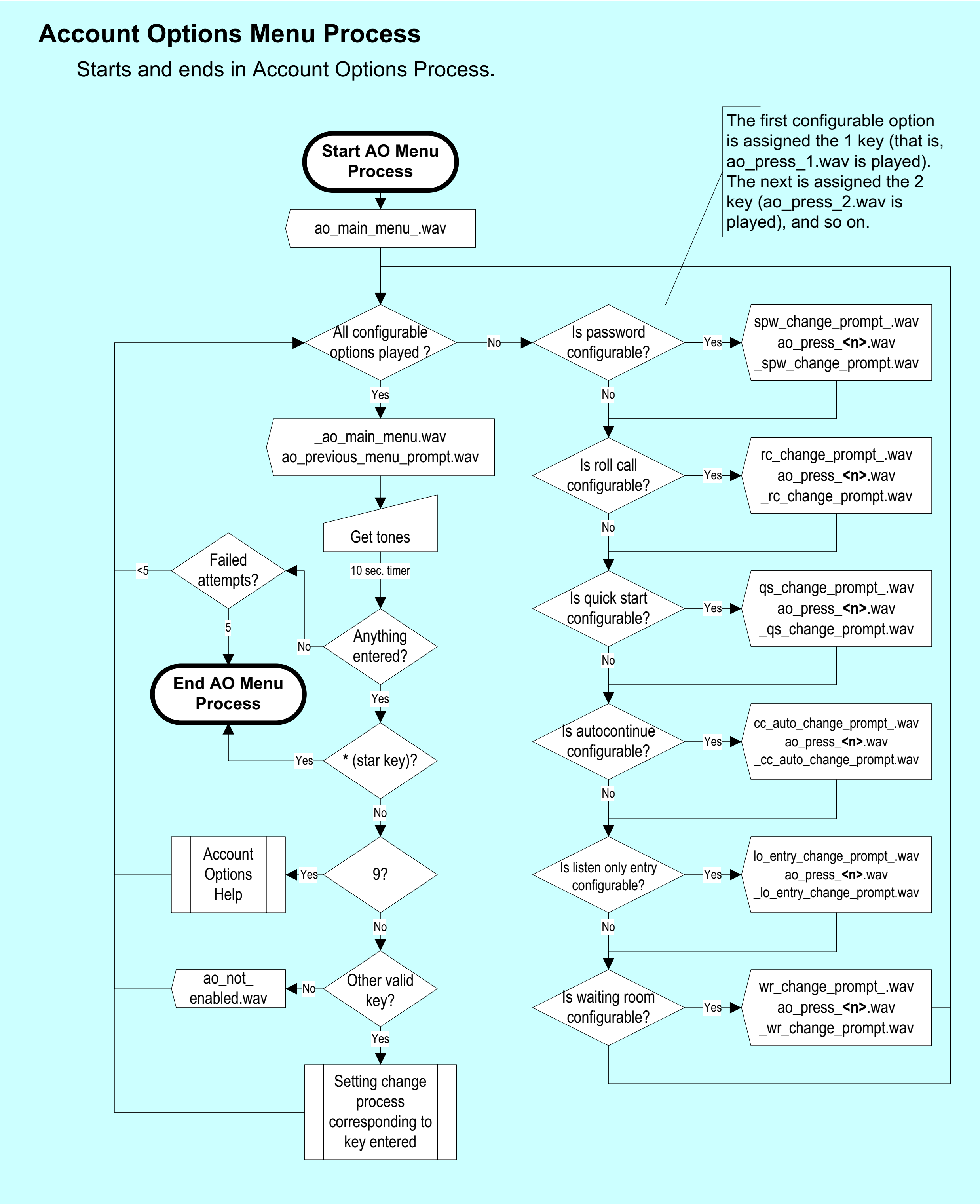


Figure A-25 *Account options help*

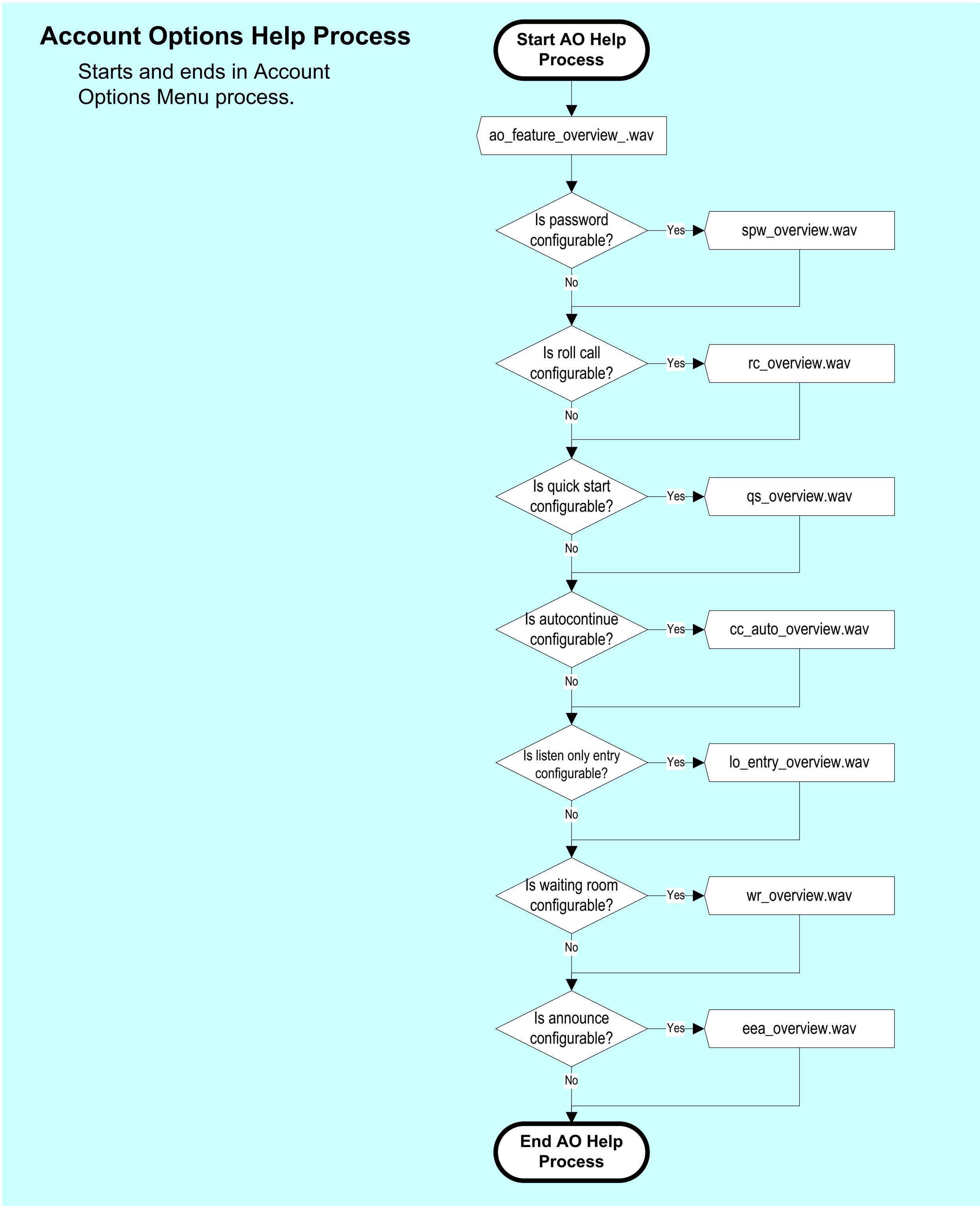


***Figure A-26** Announcement setting process*

Announcement Setting Process

Starts either from roll call setting process or from subscriber entering DTMF command (default is *3) while in conference. Ends where started.

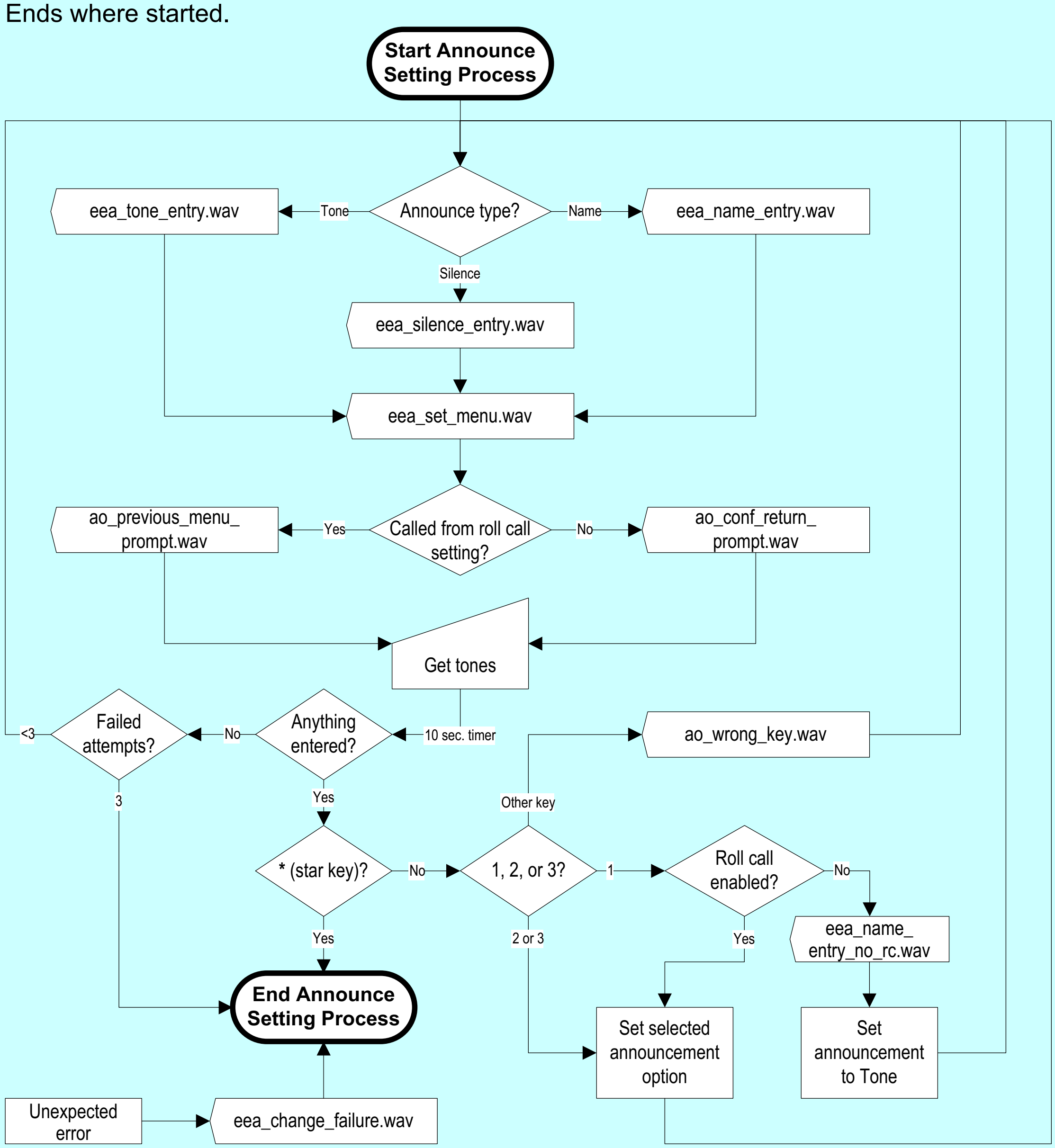


Figure A-27 *Autocontinuation setting process*

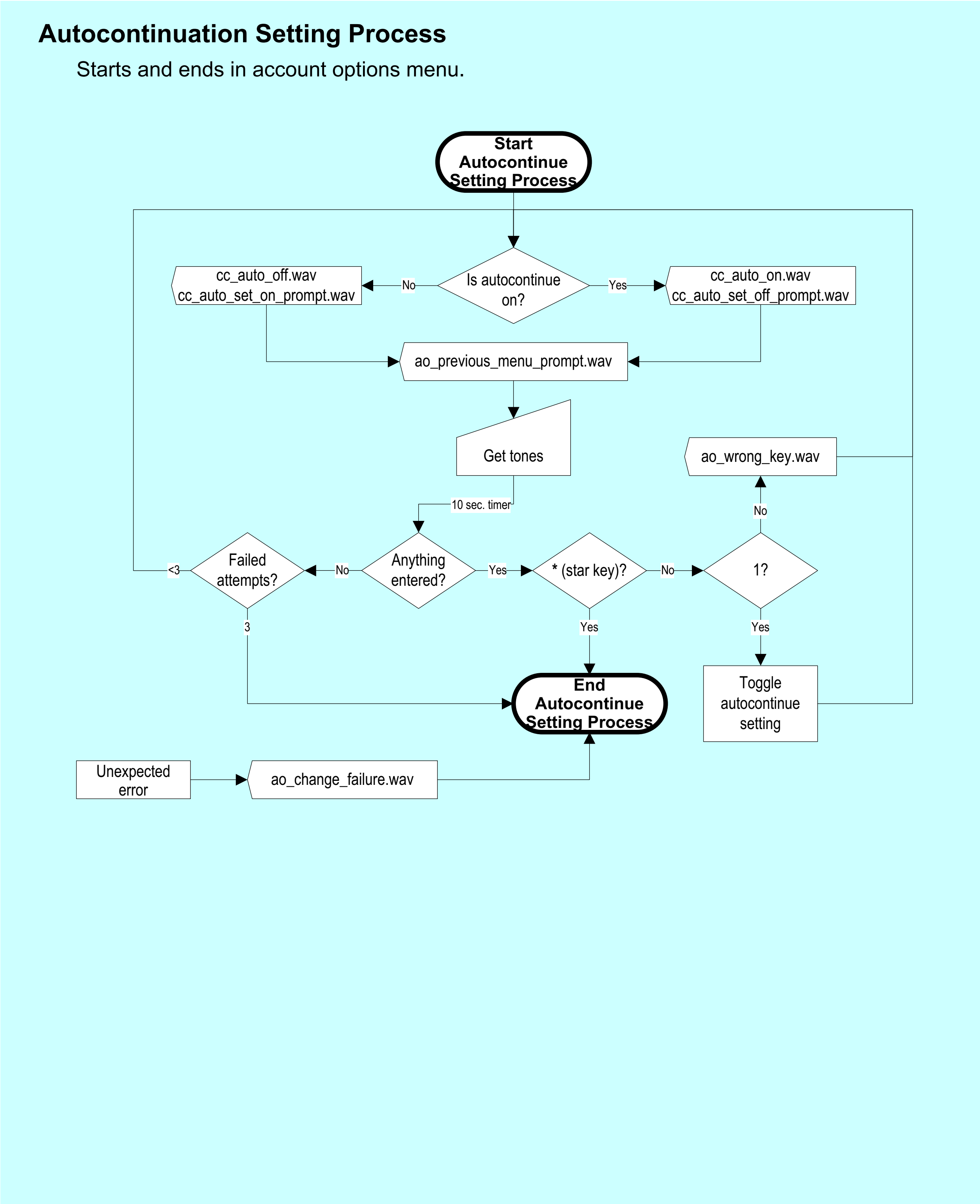


Figure A-28 *Listen only entry setting process*

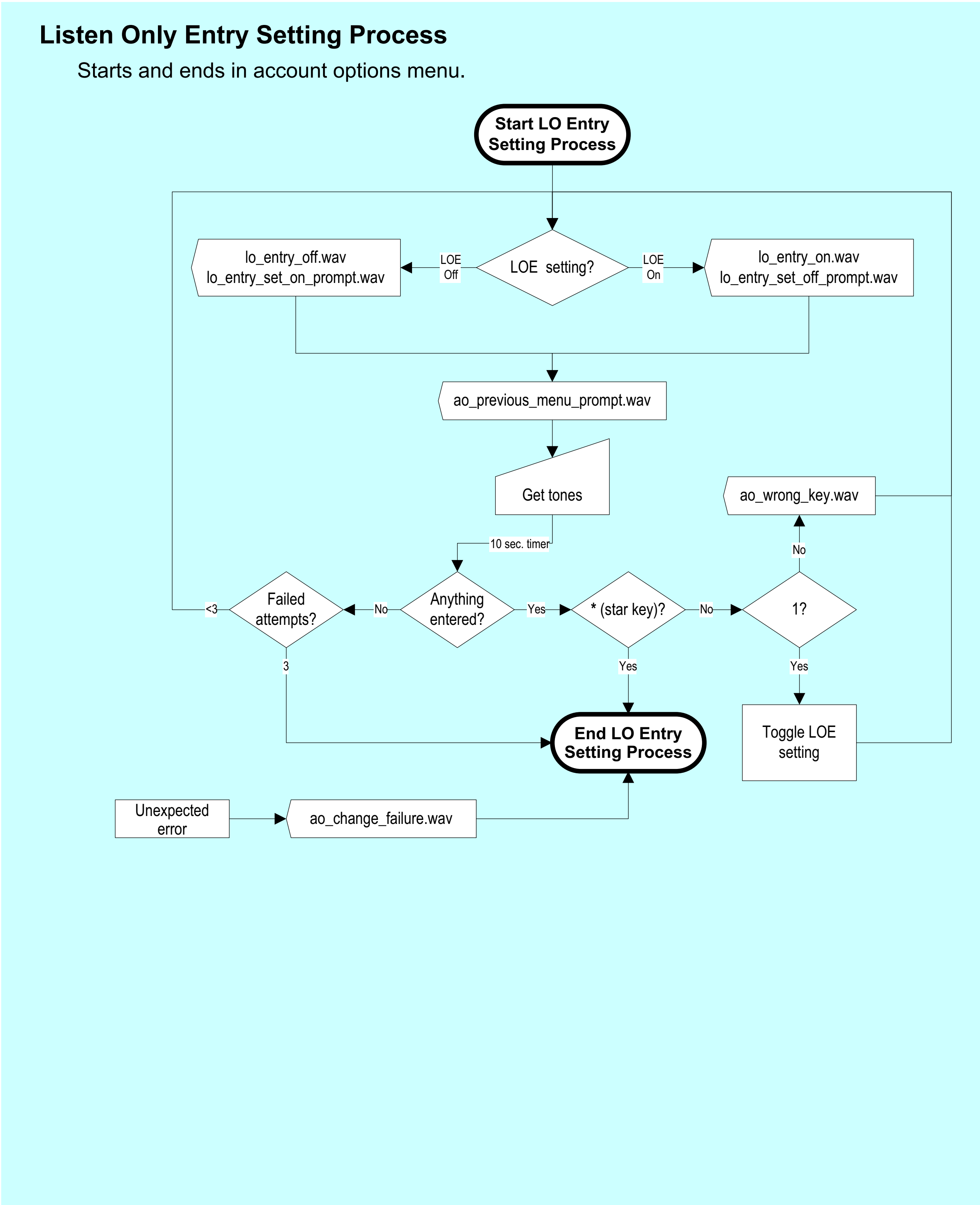


Figure A-29 Password setting process

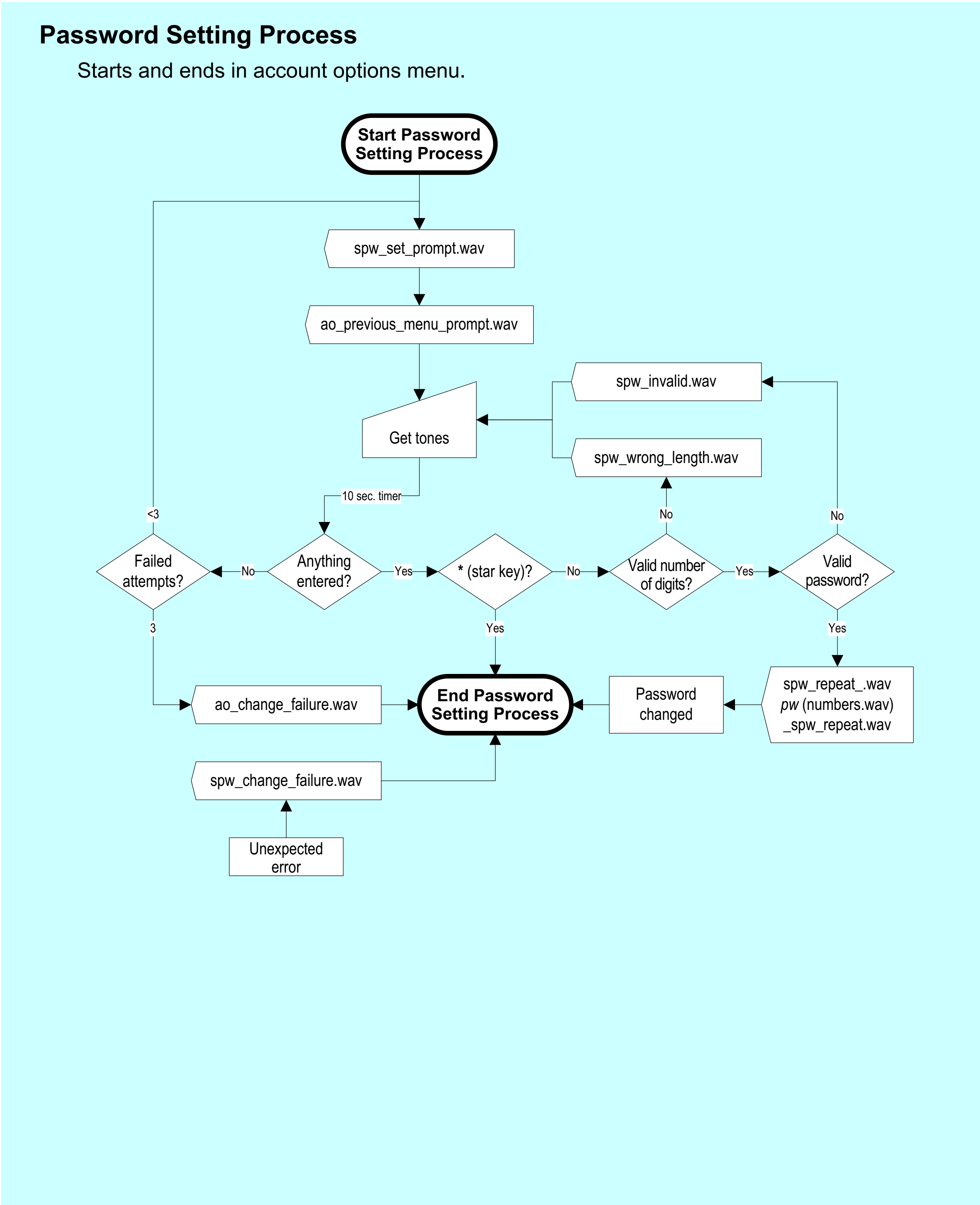


Figure A-30 *Quick start setting process*

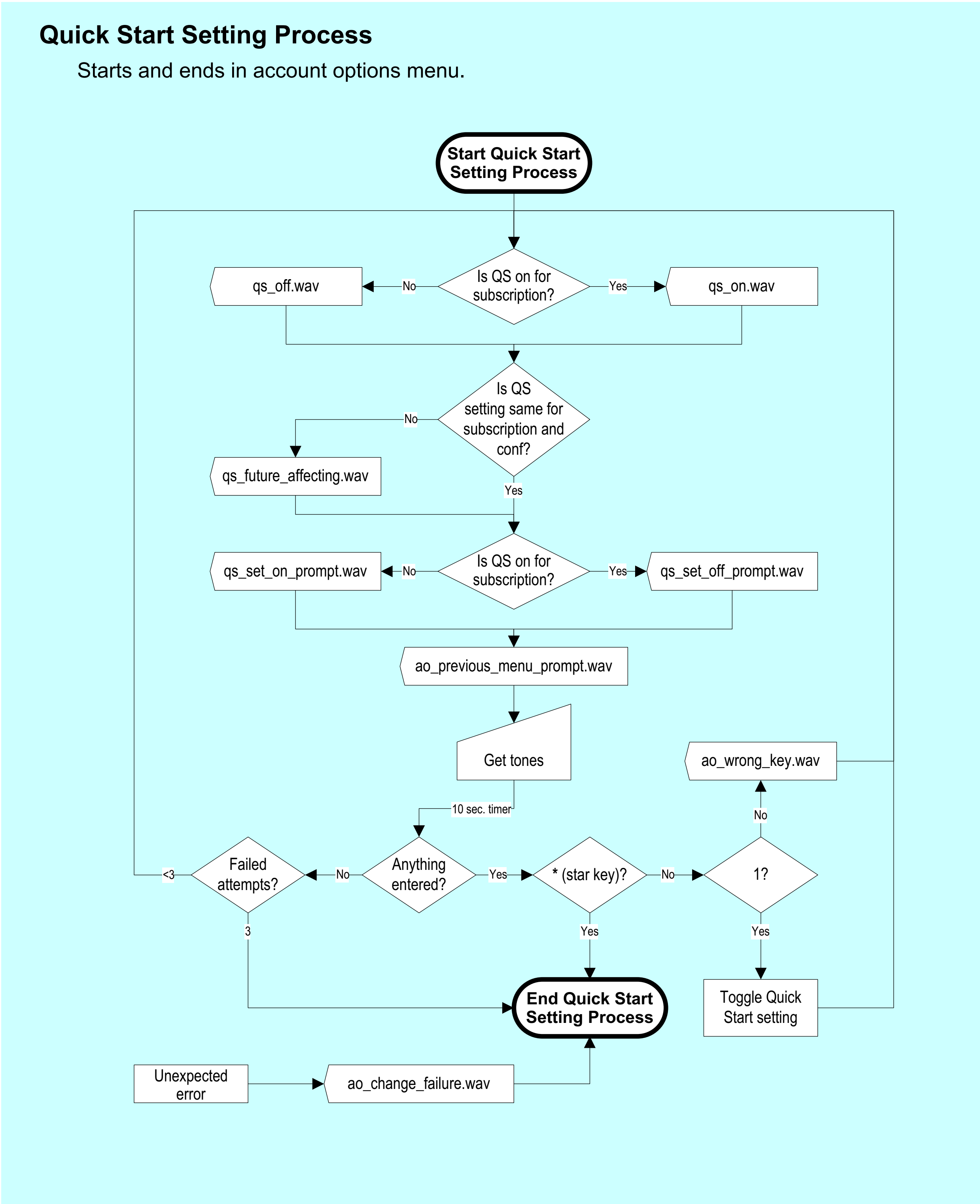


Figure A-31 *Roll call setting process*

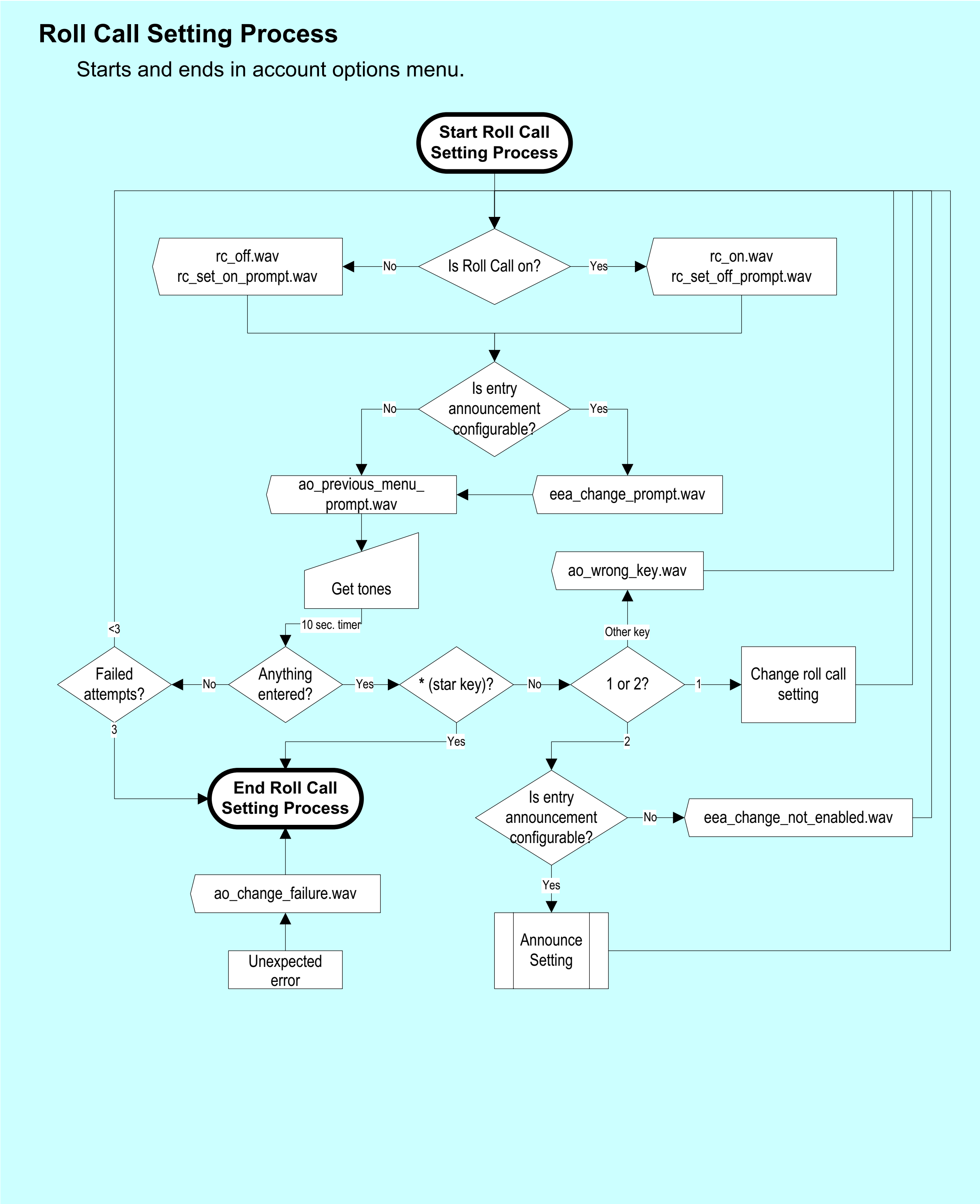


Figure A-32 Waiting room setting process

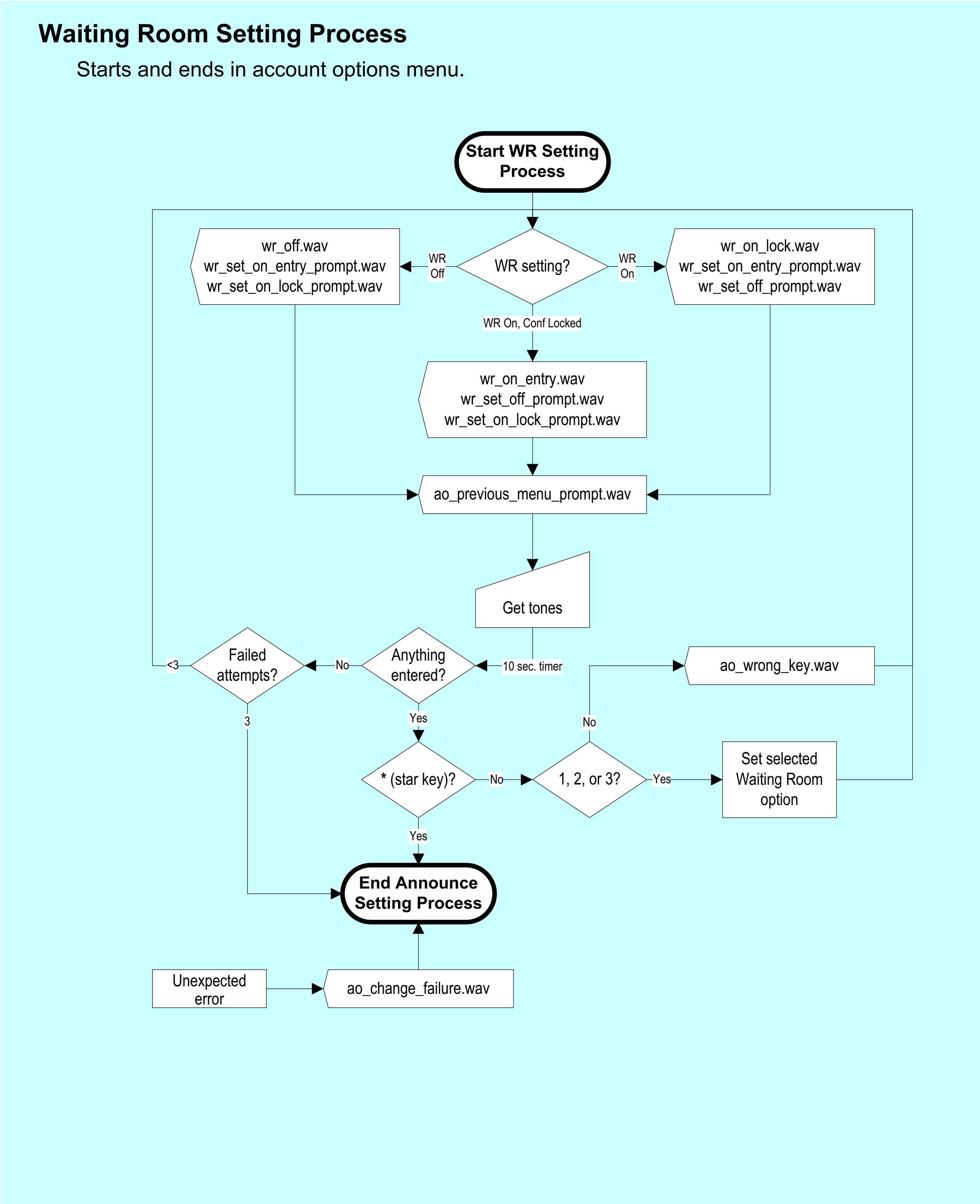


Figure A-33 *Entry announce process*

Entry Announce Process

Describes what the system plays to the conference (shaded boxes) as someone enters the conference. Starts and ends in Subscriber, Participant, or Rejoin Conference process.

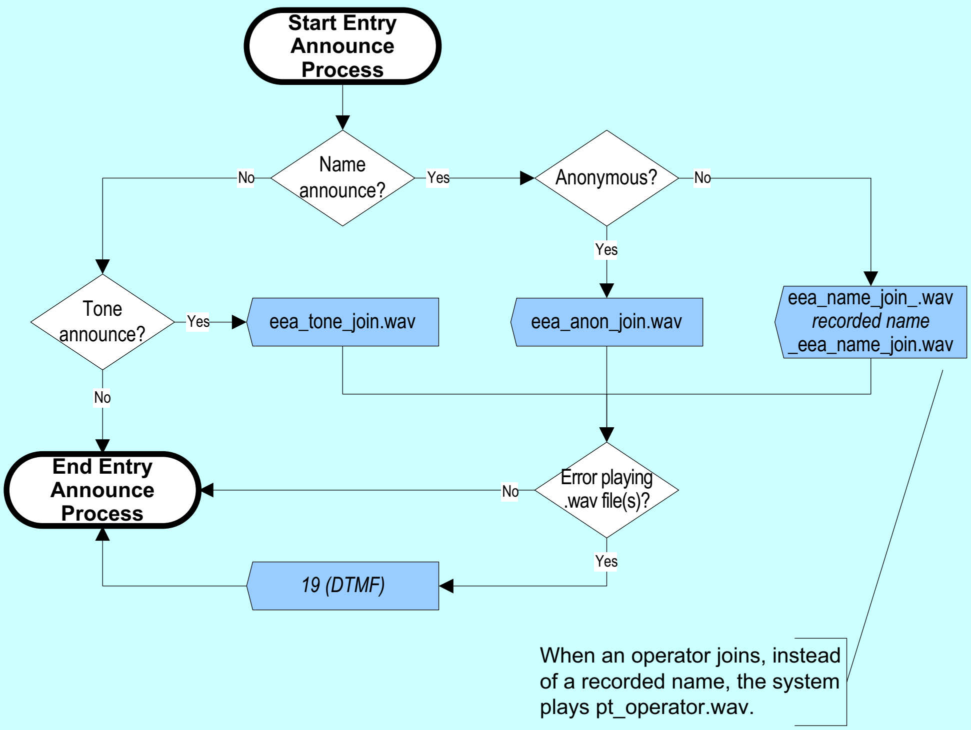


***Figure A-34** Exit announce process*

Exit Announce Process

Describes what the system plays to the conference (shaded boxes) as someone leaves the conference. Starts in conference and ends with leaving party gone (temporarily, as when subscriber departs to process WR or speak with operator, or permanently, as when someone is disconnected from the bridge).

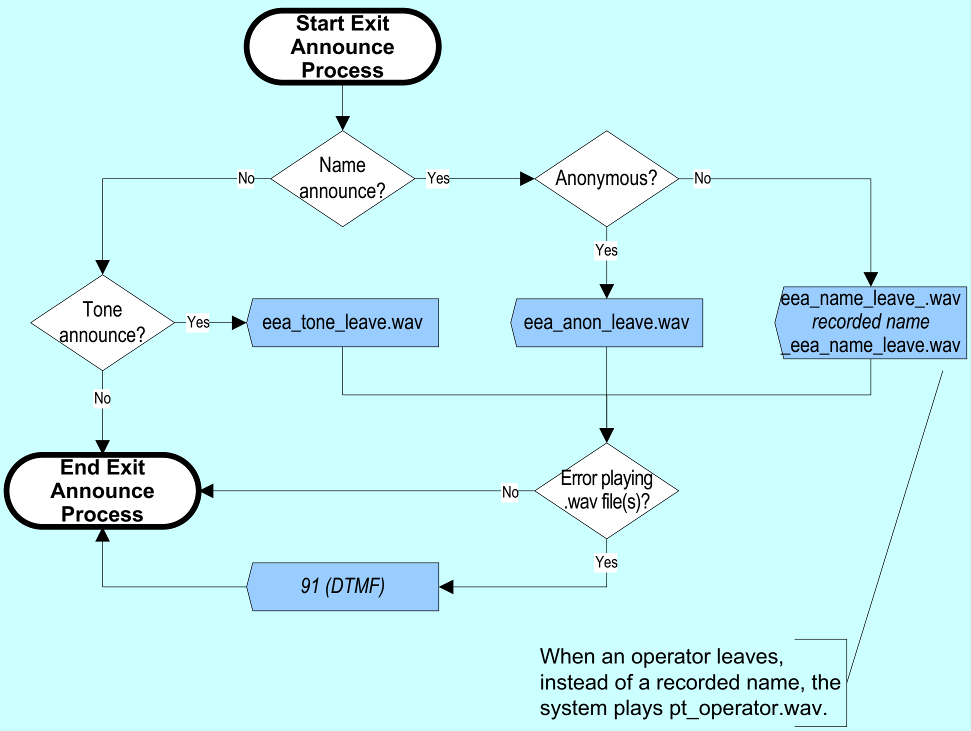


***Figure A-35** Waiting room process (subscriber)*

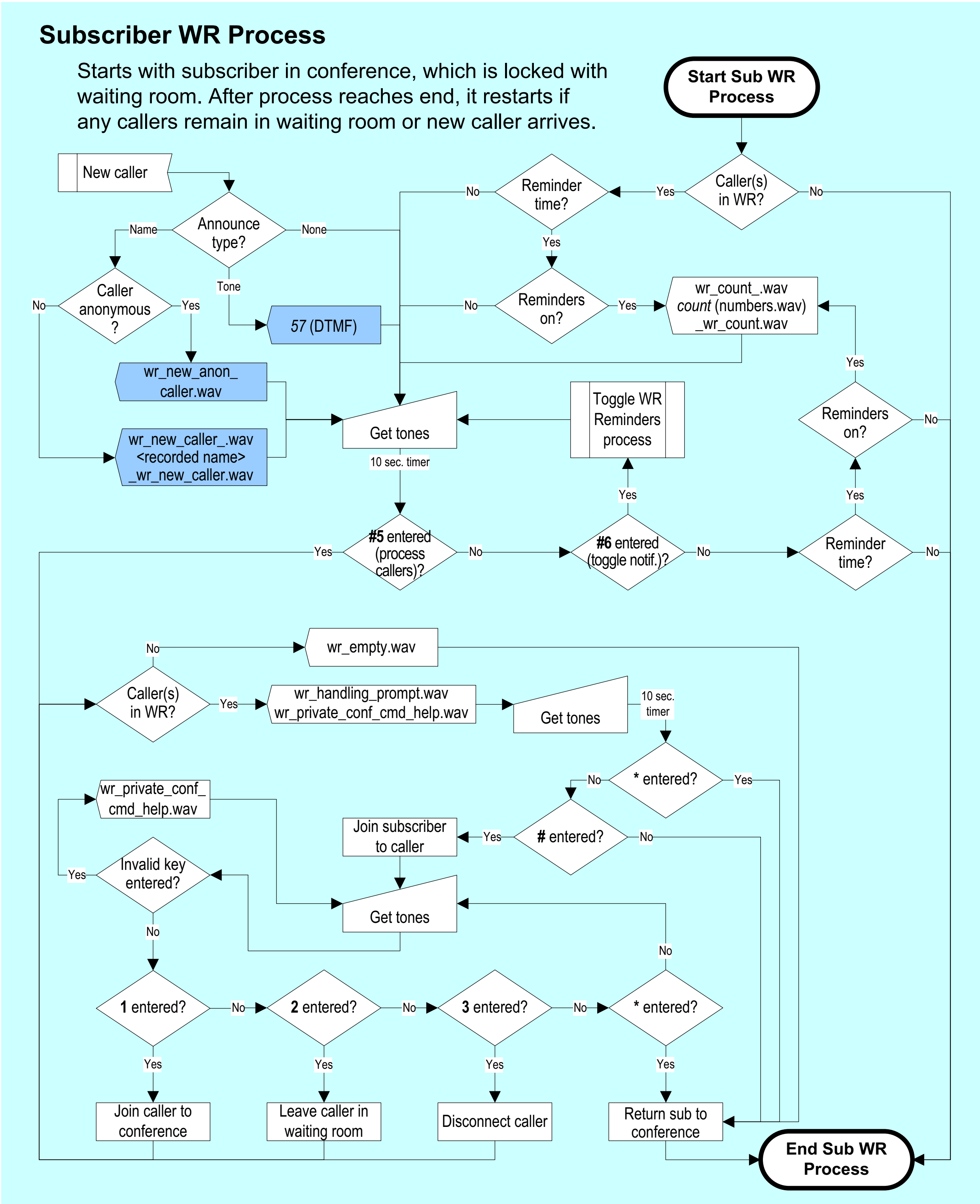


Figure A-36 *Toggle waiting room reminders process*

Toggle Waiting Room Reminders Process

Starts in conference when subscriber enters command to toggle waiting room reminders (default is #6). Ends at Rejoin Conference process.

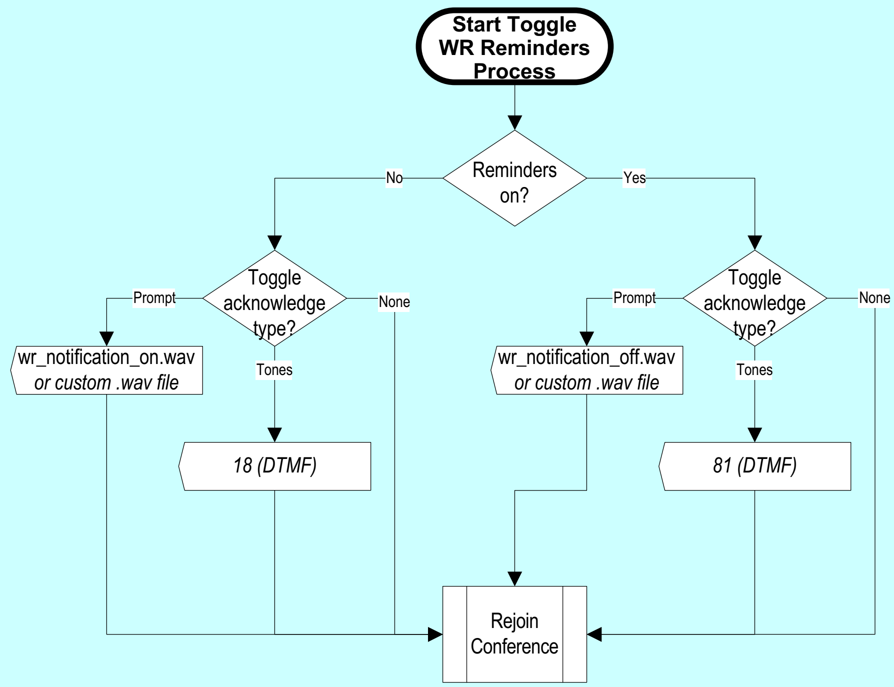


Figure A-37 *Roll call process*

Roll Call Process

Starts in conference when participant or subscriber enters roll call command.
Ends in conference or at Rejoin Conference process.

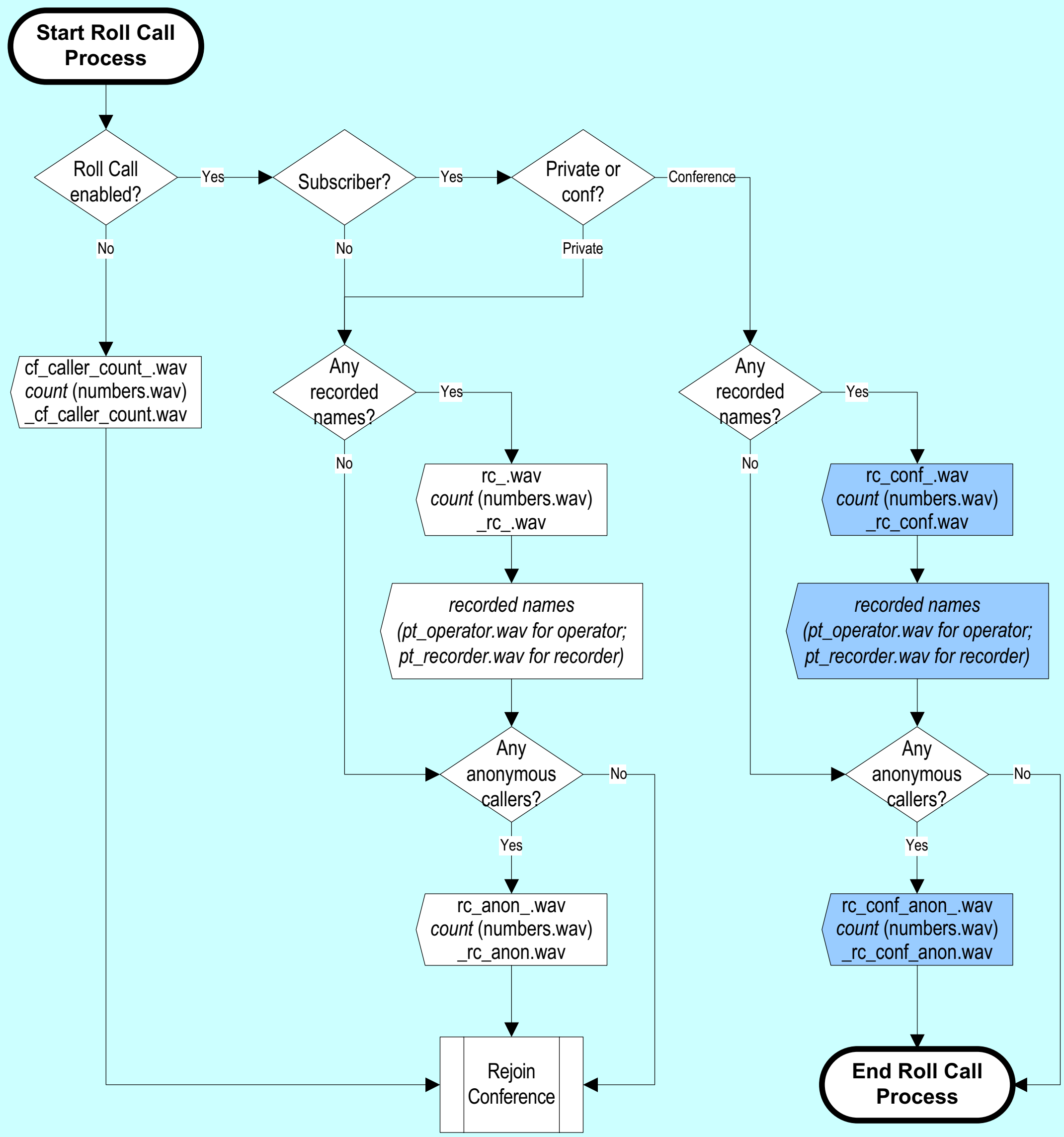


Figure A-38 *Operator request process*

Operator Request Process

Starts in conference when participant or subscriber enters operator request command (via DTMF or Moderator/API). Ends at Rejoin Conference process.

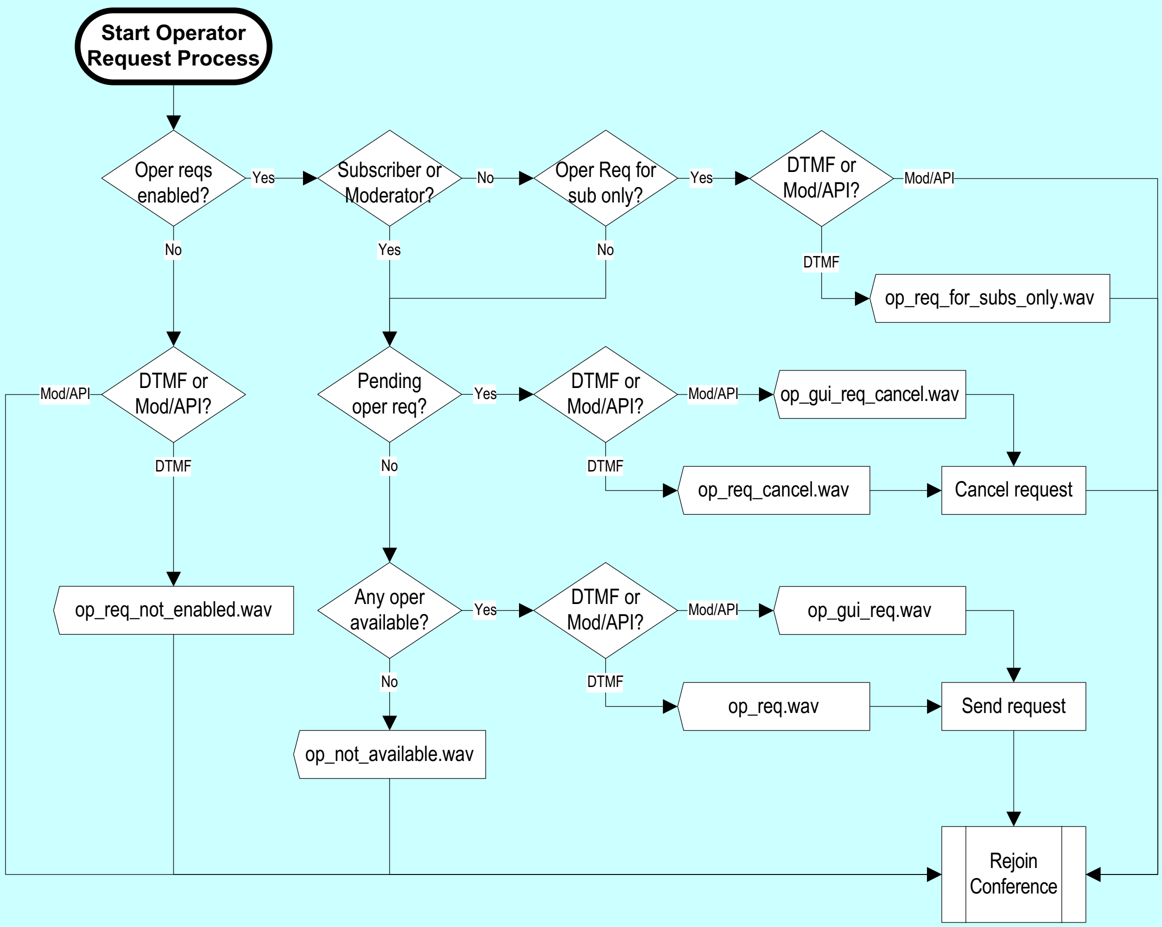


Figure A-39 Rejoin conference process

Rejoin Conference Process

When someone in conference enters a command, the system usually disconnects that person's line (channel) temporarily from the conference (although the conference can still be heard). When the requested interaction or command execution is finished, this process governs how the line is returned to the conference. It ends with the subscriber or participant back in conference and the system waiting for the next command.

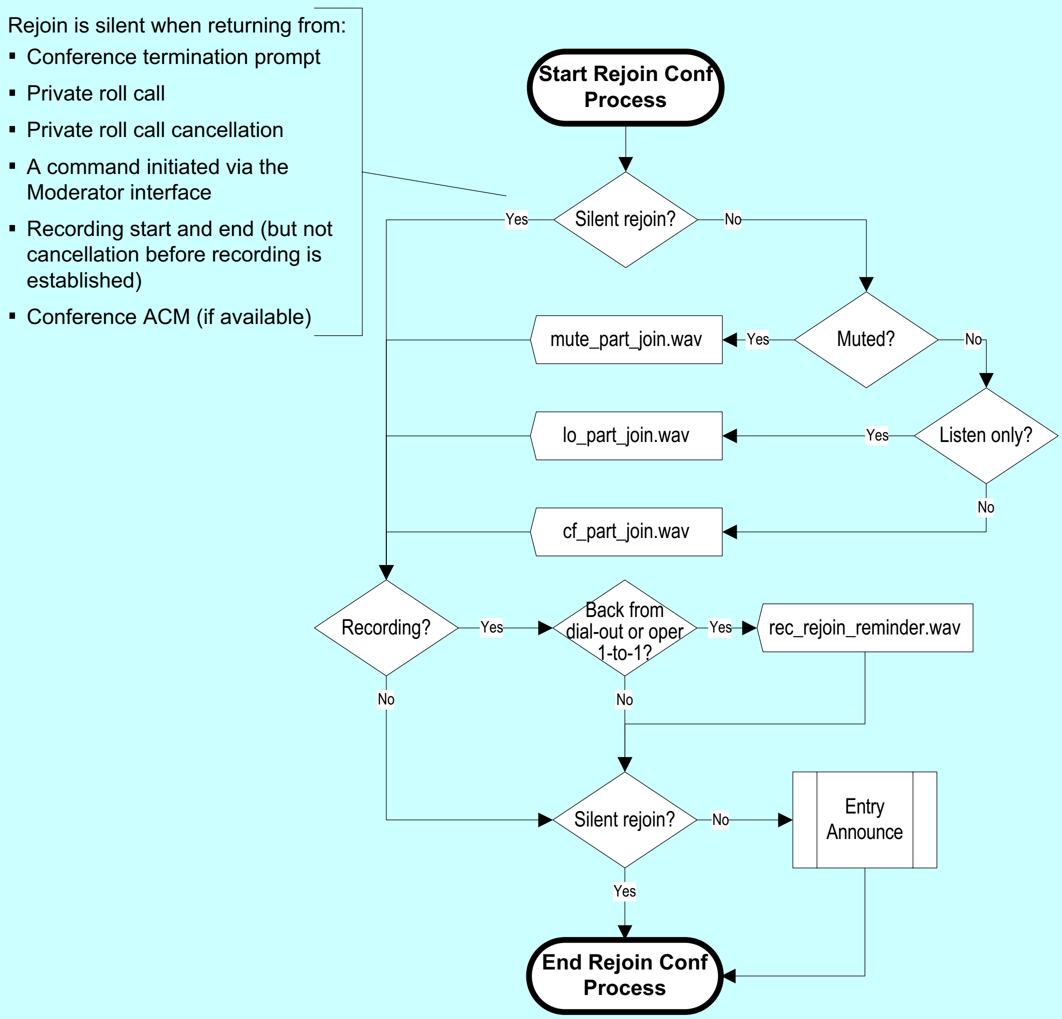


Figure A-40 *End conference process (subscriber)*

Subscriber End Conference Process

Starts in conference when subscriber enters end conference command (default is **#8**).
Ends at Rejoin Conference process or with conference termination.

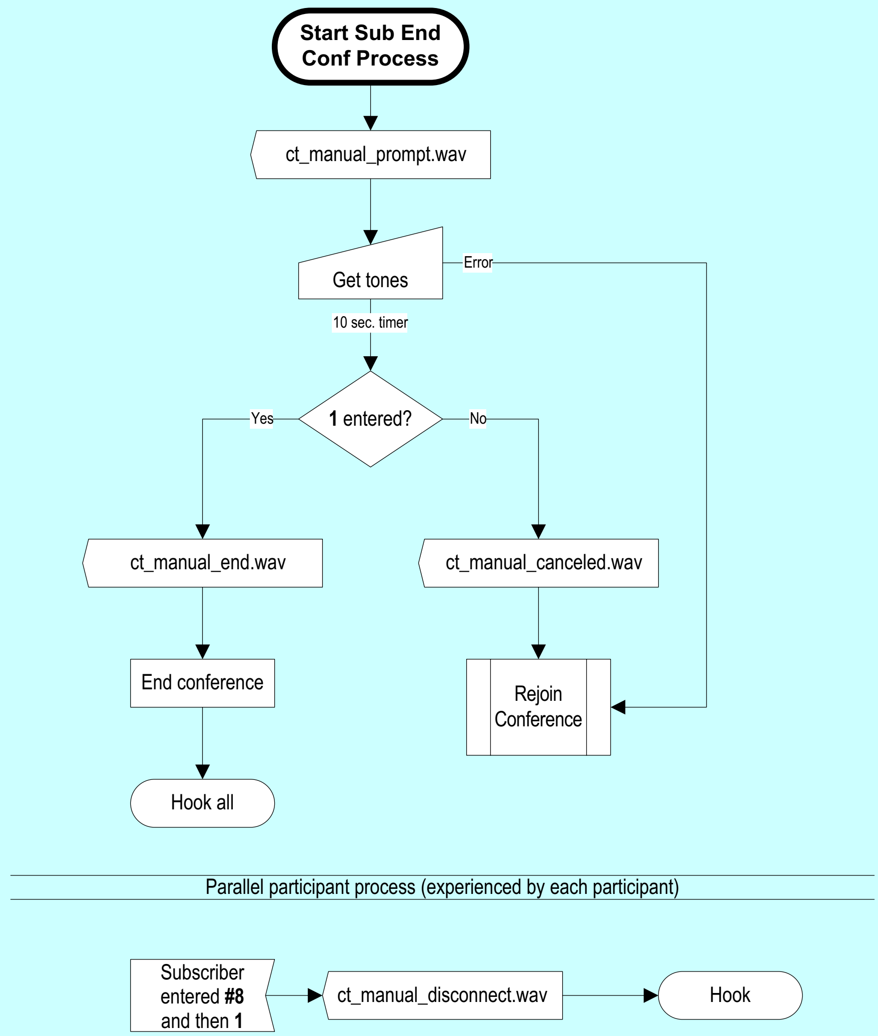


Figure A-41 *In-conference commands by subscriber*

In-conference Subscriber/Moderator Commands

All start in conference when subscriber enters indicated DTMF (touchtone) command (defaults shown) or corresponding Moderator command. Also shown here is notification of subscriber that an operator wants to join a locked conference.

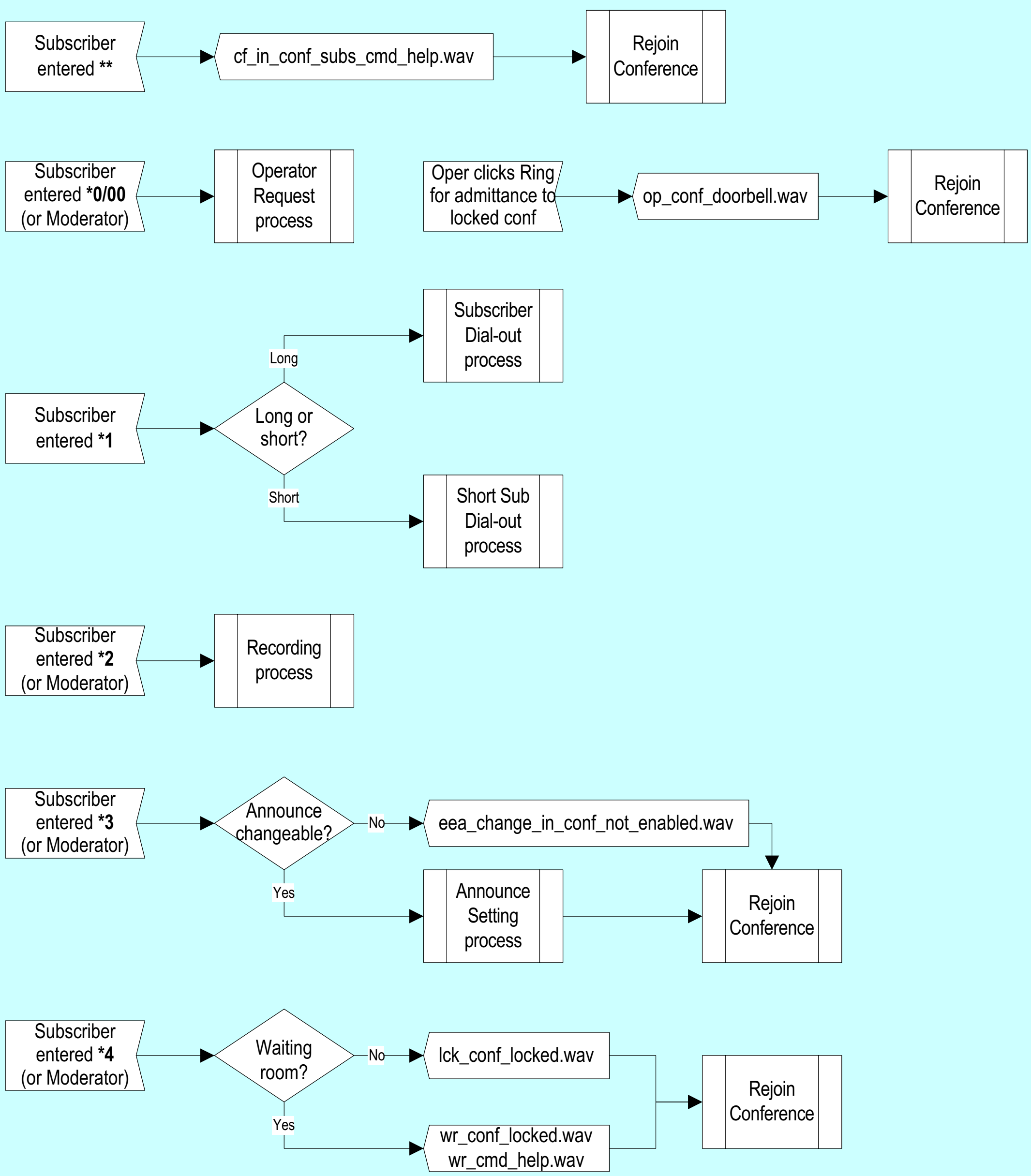


Figure A-42 *In-conference commands by subscriber (continued)*

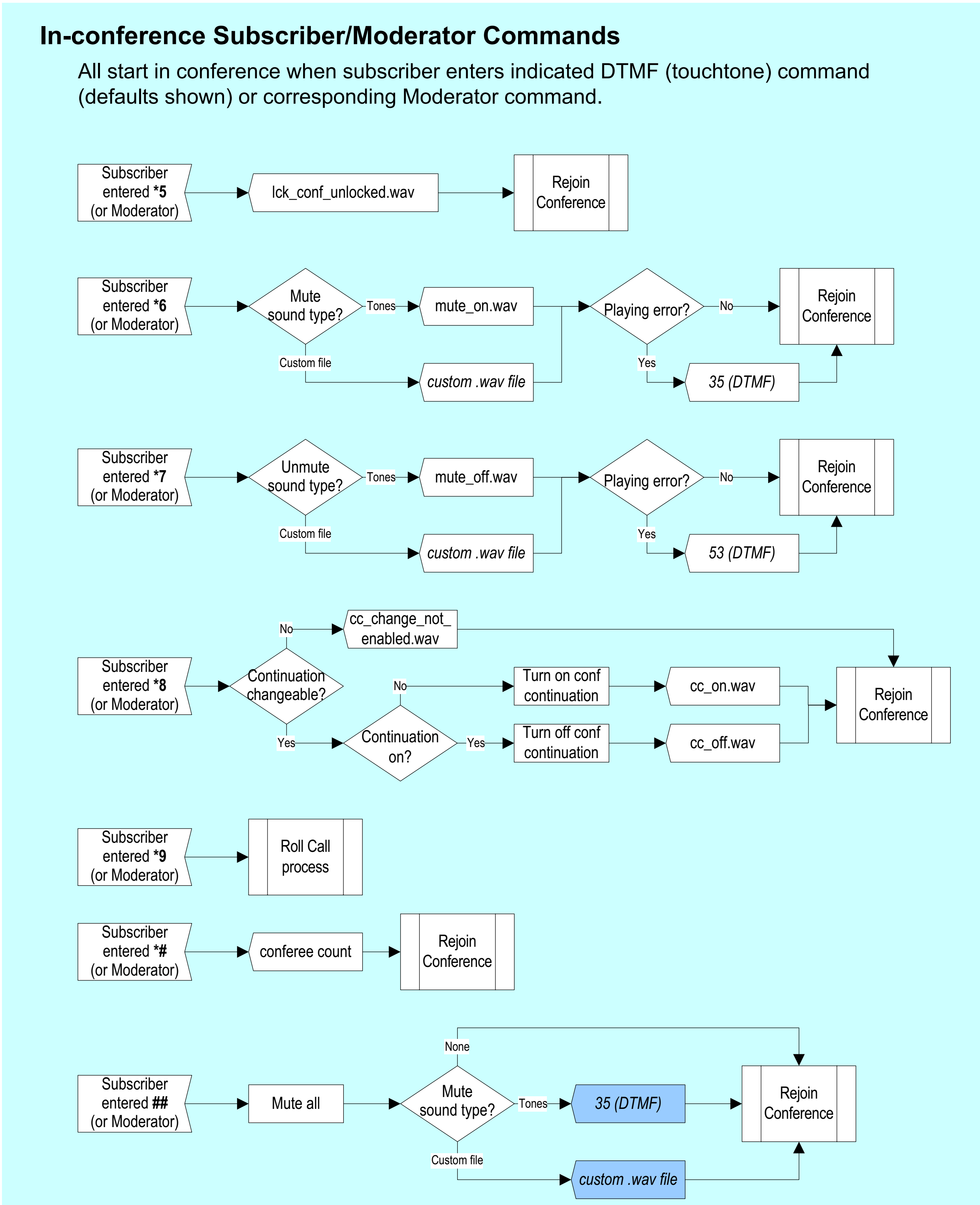


Figure A-43 *In-conference commands by subscriber (continued)*

In-conference Subscriber DTMF and Moderator Commands

All start in conference when subscriber enters indicated DTMF (touchtone) command (defaults shown) or corresponding Moderator command.

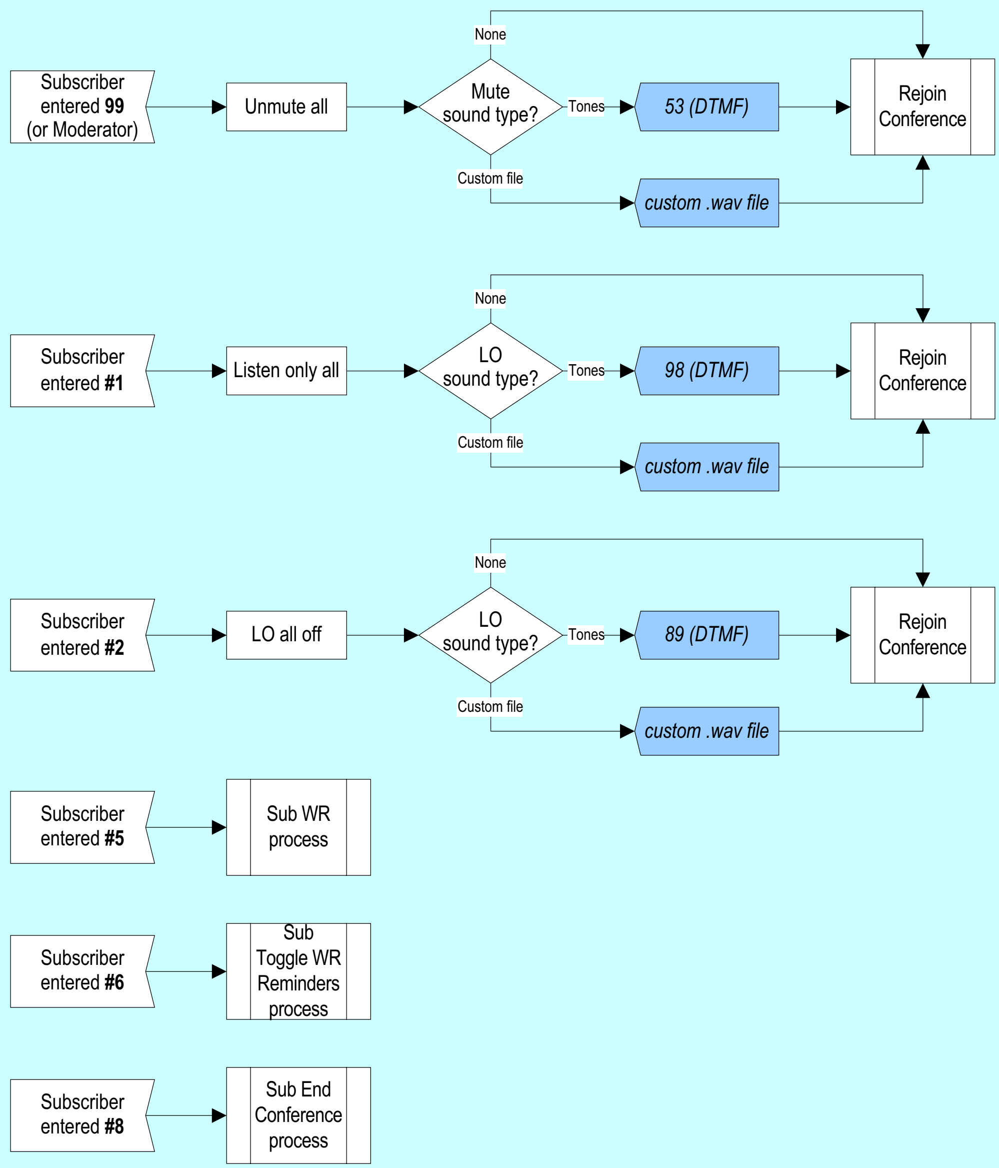


Figure A-44 *In-conference commands by participant*

In-conference Participant Commands

All start in conference when participant enters indicated DTMF (touchtone) command (defaults shown).

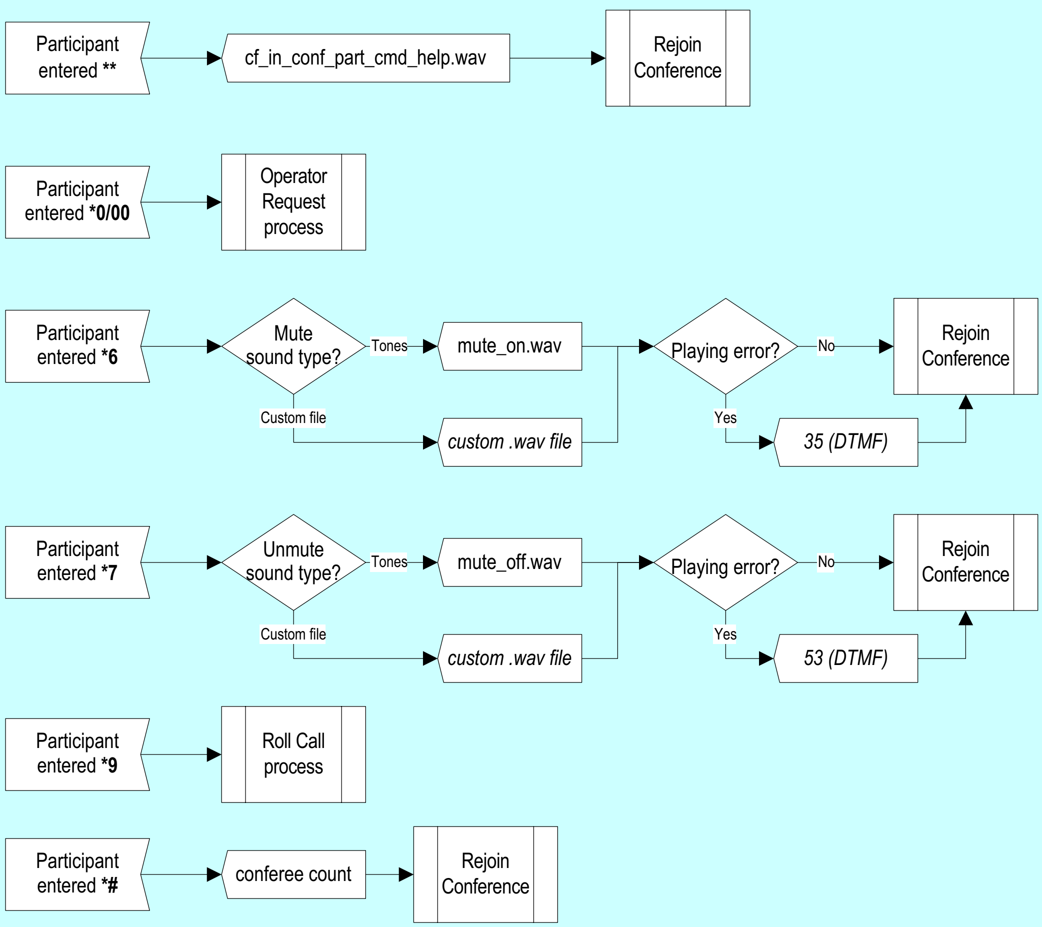


Figure A-45 Conference termination process

Conference Termination Process

Starts in conference when system determines that conference has exceeded the maximum time for a one-person (or configured minimum) conference. Ends at Rejoin Conference process or with conference termination.

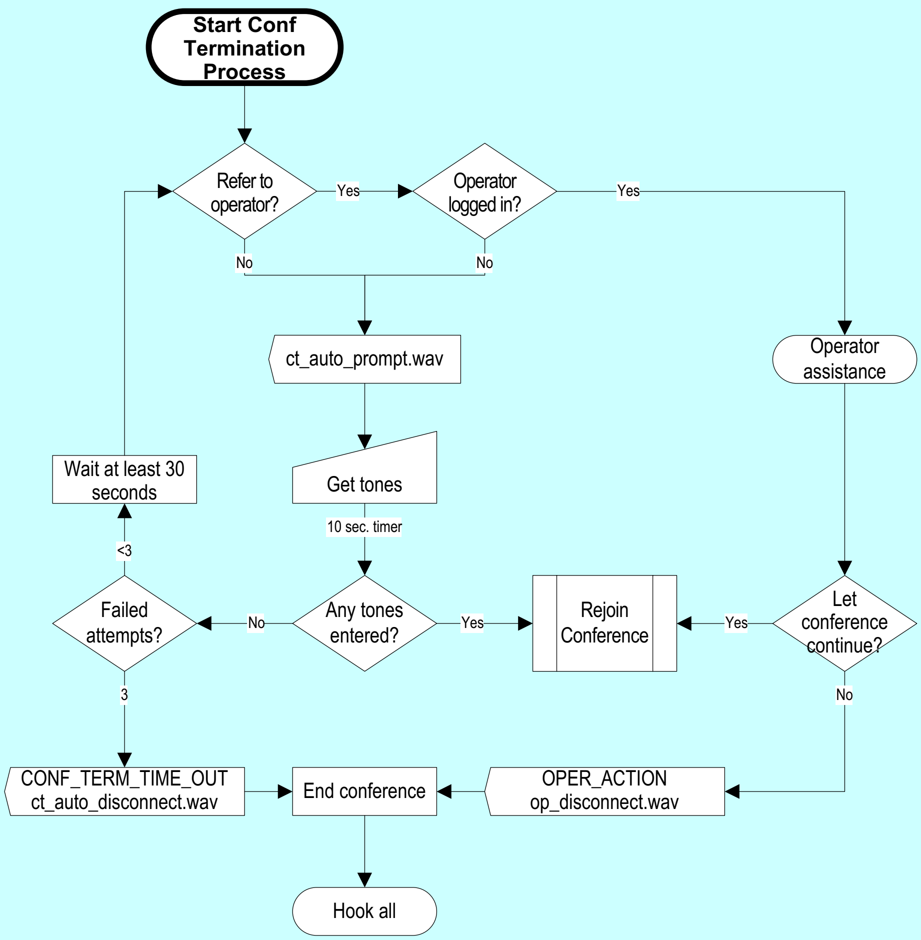
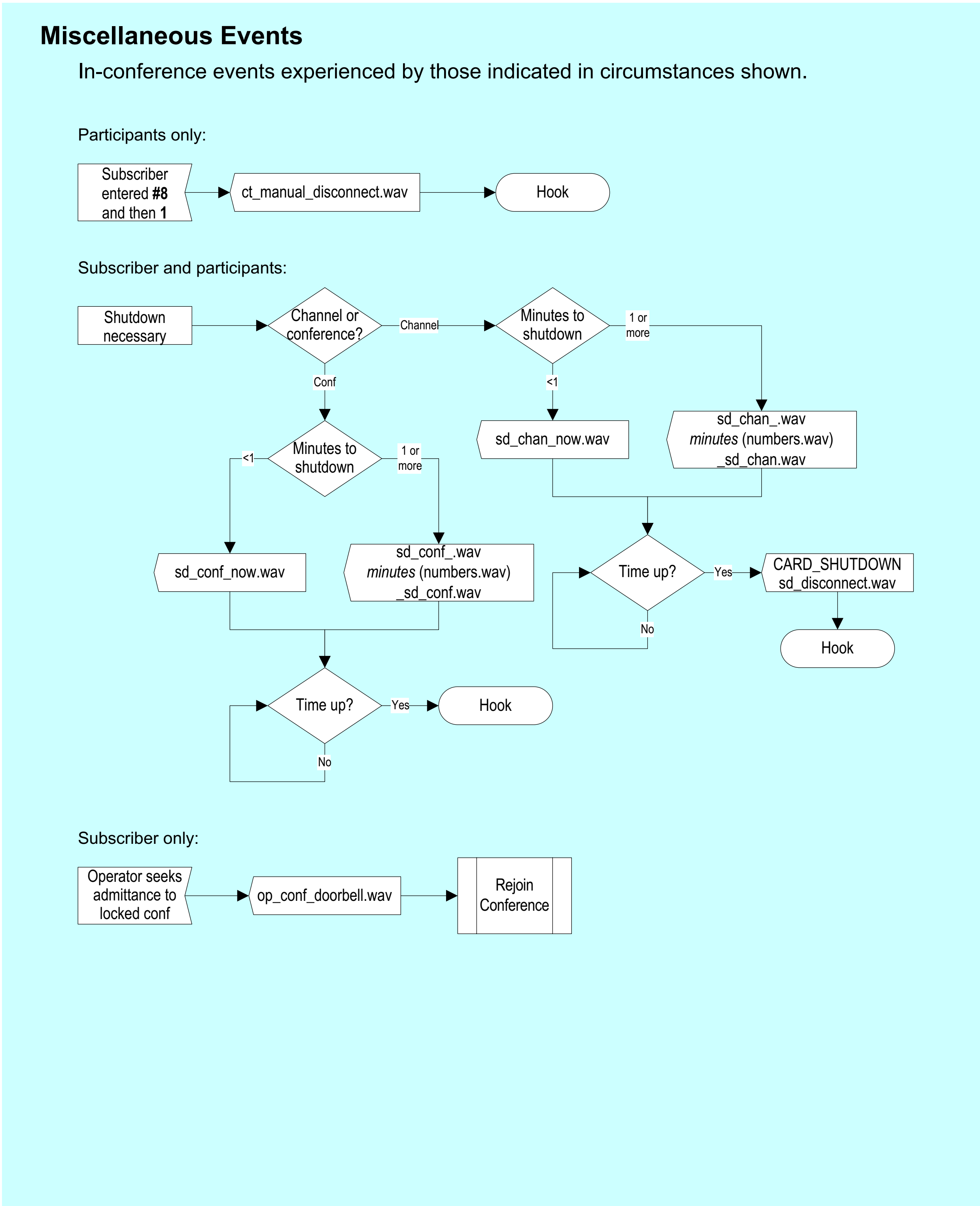


Figure A-46 *Other in-conference events*



B

CDR Data Reference

This appendix describes post-conference data found in the Call Detail Records (CDRs). CDRs provide the billing information for your ReadiVoice system.

CDR Processing Database

The CDR database is composed of five Informix tables. These tables are kept in a DBSPACE within the Informix storage area. By convention, this DBSPACE is called dbspace1. Since the storage of CDR information is not a core system component (that is, not necessary for conference functionality), it's designed to have minimal impact on other Conference Allocation and Control System (CACS) components. This includes all Informix references and queries.

Informix CDR Tables

Five Informix tables hold information from completed conferences:

- Conference Information (`cdr_post_conf`)
- Participant Information (`cdr_post_part`)
- Feature Information (`cdr_post_state`)
- Conference End Verification (`cdr_end_conf`)
- ACM Data (`cdr_acm_data`)

The tables that follow describe the contents of each Informix table. Fields in bold type have indexes.

Table B-1 *Conference Information (cdr_post_conf)*

Field Name	Field Type	Description
customer_id	INTEGER	Unique billing ID or customer reference number. In the CNOW database, this is SUBSCRIBERID from the SUBSCRIBERINFO table.
conf_id	INTEGER	Unique conference ID number.
sub_id	SMALLINT	Forced to zero (0).
resv_id	INTEGER	In the CNOW database, this is GROUPID from the SUBSCRIBERINFO table.
chairperson_name	CHAR(30)	Not applicable.
conf_actual_start	INTEGER	Start time of conference in UNIX seconds.
conf_actual_end	INTEGER	End time of conference in UNIX seconds.
service_provider	CHAR(20)	In the CNOW database, this is CARRIERNAME from the CARRIERS table.
reservation_op_id	CHAR(6)	Not applicable.
participant_num	SMALLINT	Number of actual participants who called into the conference.
voting_duration	SMALLINT	Forced to zero (0).
q_a_duration	SMALLINT	Forced to zero (0).
processed	SMALLINT	ReadiVoice sets this flag to zero (0). The ReadiVoice CDR purge program will delete CDRs that have this flag set to 1 by CDR fetching programs.

Table B-1 Conference Information (cdr_post_conf) (continued)

Field Name	Field Type	Description
reserver_title	CHAR(10)	Title of subscriber. In the CNOW database, this is TITLE from the SUBSCRIBERDETAIL table.
reserver_first	CHAR(30)	First name of subscriber. In the CNOW database, this is FIRSTNAME from the SUBSCRIBERINFO table.
reserver_last	CHAR(30)	Last name of subscriber. In the CNOW database, this is LASTNAME from the SUBSCRIBERINFO table.
reserver_phone	CHAR(30)	Phone number of subscriber. In the CNOW database, this is PHONE from the SUBSCRIBERDETAIL table.
resv_begin	INTEGER	Forced to zero (0).
resv_end	INTEGER	Forced to zero (0).
resv_lines	SMALLINT	Number of lines authorized for subscription.
conf_type_code	CHAR(4)	Forced to "AUMM".
oper_notes	CHAR(55)	Not applicable.
adver_code	CHAR(15)	Not applicable.
billing_id	INTEGER	In the CNOW database, this is BILLINGID from the SUBSCRIBERINFO table.
r_auto_lo_conf	SMALLINT	Forced to zero (0).
r_fax_confirm	SMALLINT	Forced to zero (0).
r_mon_100	SMALLINT	Forced to zero (0).
r_announce	SMALLINT	Forced to zero (0).
r_announce_late	SMALLINT	Forced to zero (0).
r_music	SMALLINT	Forced to zero (0).
r_roll_call	SMALLINT	Forced to zero (0).
r_smart_poll	SMALLINT	Forced to zero (0).
r_operator_tone	SMALLINT	Forced to zero (0).
r_conf_record	SMALLINT	Forced to zero (0).
r_auto_dial	SMALLINT	Forced to zero (0).

Table B-1 *Conference Information (cdr_post_conf) (continued)*

<table>
<tr><th>Field Name</th><th>Field Type</th><th colspan="2">Description</th></tr>
<tr><td>r_security_enable</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>r_pre_notify</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>r_full_duplex</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>u_no_show</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>u_cancelled</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>externalid_b</td><td>CHAR(30)</td><td colspan="2">Optional External ID B (an alternative ID) for subscriber. In the CNOW database, this is ExternalIdB from the SUBSCRIBERINFO table.</td></tr>
<tr><td>bridge_id</td><td>INTEGER</td><td colspan="2">Identifies the bridge on which the conference was running.</td></tr>
<tr><td>card_num</td><td>SMALLINT</td><td colspan="2">Identifies the card on which the conference was running.</td></tr>
<tr><td rowspan="5">cdr_status</td><td rowspan="5">CHAR(1)</td><td colspan="2">Identifies the status of the CDR. Possible values:</td></tr>
<tr><td>‘ ‘</td><td>The conference ended normally.</td></tr>
<tr><td>‘?‘</td><td>The CDR is incomplete (i.e., the conference is still active).</td></tr>
<tr><td>‘H‘</td><td>The conference ended abnormally (e.g., the card died).</td></tr>
<tr><td>‘A‘</td><td>The CDR was cleaned by a CDR cleaner (when the card running the conference dies all of the calls from that conference will have this status unless they already had a status of ‘H‘ or ‘ ‘).</td></tr>
</table>

Table B-2 *Participant Information (cdr_post_part)*

Field Name	Field Type	Description
conf_id	INTEGER	Conference ID (key to cdr_post_conf table) (indexed with part_id).
sub_id	SMALLINT	Forced to zero (0).
subscriber_number	CHAR(20)	Dial-in = A number Dial-out = B number
connect_number	CHAR(20)	B number (DNIS).
billing_number	CHAR(20)	Not applicable.
part_actual_start	INTEGER	Start time for participant in UNIX seconds (when participant connects to bridge).
part_actual_end	INTEGER	End time for participant in UNIX seconds (when participant disconnects from bridge).
access_method	SMALLINT	Enum: 0 = ACC_CAMM (subscriber dial-in). 1 = ACC_PAMM (participant dial-in). 4 = ACC_COMM (operator-initiated). 6 = ACC_CODS (subscriber dial-out). 7 = ACC_PODS (participant dial-out).
part_id	INTEGER	Participant ID number (indexed alone and with conf_id).
part_privacy	SMALLINT	Forced to zero (0).
part_privilege	SMALLINT	Enum: 0 = ordinary participant 1 = subscriber 3 = operator 4 = recorder
part_title	CHAR(10)	If subscriber record, the subscriber's TITLE from the SUBSCRIBERDETAIL table (CNOW database). If participant, operator, or recorder record, blank.

Table B-2 *Participant Information (cdr_post_part) (continued)*

Field Name	Field Type	Description
part_first	CHAR(30)	If subscriber record, one of the following: • ACM Pins value, if any • Blank, if participant name updated (via API, Operator, or Moderator) and no ACM Pins • Otherwise, FIRSTNAME from SUBSCRIBERINFO table If participant record, one of the following: • ACM Pins value, if any • Blank, if participant name updated (via API, Operator, or Moderator) and no ACM Pins • Otherwise, “Participant” If operator or recorder record, “Operator” or “Recorder,” respectively.
part_last	CHAR(30)	If subscriber record, one of the following: • Updated name, if participant name updated (via API, Operator, or Moderator) • Otherwise, LASTNAME from SUBSCRIBERINFO table If participant record, one of the following: • Updated name, if participant name updated (via API, Operator, or Moderator) and no ACM Pins • Otherwise, blank If operator or recorder record, blank.
bridge_id	INTEGER	Bridge ID of bridge to which participant connected.
bridge_leg_id	SMALLINT	Forced to 1.
leg_type	SMALLINT	Forced to zero (0).
gmt_offset	SMALLINT	Forced to zero (0).
in_out_flag	SMALLINT	Enum: 0 = dial-in 1 = dial-out
r_op_assist	SMALLINT	Forced to zero (0).
r_price_req	SMALLINT	Forced to zero (0).
r_person_person	SMALLINT	Forced to zero (0).
r_listen_only	SMALLINT	Forced to zero (0).

Table B-2 *Participant Information (cdr_post_part) (continued)*

<table>
<tr><th>Field Name</th><th>Field Type</th><th colspan="2">Description</th></tr>
<tr><td>port_group</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>res_port_group</td><td>SMALLINT</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>line_number</td><td>INTEGER</td><td colspan="2">Forced to zero (0).</td></tr>
<tr><td>card_num</td><td>SMALLINT</td><td colspan="2">Identifies the card on which the conference was running.</td></tr>
<tr><td rowspan="5">cdr_status</td><td rowspan="5">CHAR(1)</td><td colspan="2">Identifies the status of the CDR. Possible values:</td></tr>
<tr><td>‘ ‘</td><td>The call ended normally.</td></tr>
<tr><td>‘?‘</td><td>The CDR is incomplete (i.e., the call is still active).</td></tr>
<tr><td>‘H‘</td><td>The call ended abnormally (e.g., the card died).</td></tr>
<tr><td>‘A‘</td><td>The CDR was cleaned by a CDR cleaner (when the card running the conference dies all of the calls from that conference will have this status unless they already had a status of ‘H‘ or ‘ ‘).</td></tr>
</table>

Table B-3 *Feature Information (cdr_post_state)*

Field Name	Field Type	Description
`conf_id`	`INTEGER`	Conference ID (key to cdr_post_conf table).
`part_id`	`INTEGER`	Participant ID (key to cdr_post_part table).
`feature`	`SMALLINT`	Type of feature used: **Duration Events**[a] 8 = `oper_request` (operator request – private) 9 = `oper_confreq` (operator request – conference) 10 = `dial_out` (subscriber dial-out) 11 = `conf_lock` (conference locked) 15 = `conf_record` (conference recorded) 21 = `chan_in_pre_acm` (channel in pre-conf ACM) 22 = `chan_in_ic_acm` (channel in in-conf ACM) **Common Events**[b] 4 = `chan_join` (someone joined the conference) 12 = `chan_mute` (channel muted) 13 = `chan_unmute` (channel unmuted) 14 = `virt_gavel` (conference muted) 17 = `roll_call` (roll call enabled) 18 = `conf_continue` (conference continuation enabled) 19 = `quick_start` (Quick Start conference) 20 = `stream` (not supported) 23 = `op_req` (operator request – private) 24 = `op_confreq` (operator request – conference) 25 = `cancel_req` (cancel operator request) 28 = `listen_only` (channel is in listen only mode) 29 = `chan_in_wr` (channel is in waiting room) **Reserved Events** 0-3, 5-7, 16, 26, 27

Table B-3 Feature Information (cdr_post_state) (continued)

Field Name	Field Type	Description
`oper_id`	`INTEGER`	Participant ID of the operator answering the request. Set for duration events 8 and 9; otherwise N/A.
`start_time`	`INTEGER`	Start timestamp of feature in UNIX seconds.
`end_time`	`INTEGER`	End timestamp of feature in UNIX seconds (set to 0 for common events).

a. Duration events have both a start_time and end_time.
b. Common events require a start_time; end_time is always zero (0).

Table B-4 Conference End Verification (cdr_end_conf)

Field Name	Field Type	Description
`conf_id`	`INTEGER`	Conference ID of conference that has all of its CDRs stored in the database.
`water_mark`	`SERIAL`	Available for use as a water mark for processing.

The ACM Data table is available for storing additional data collected by ACM (Application Control Mode) applications that you develop. The ACM application is responsible for sending the collected data to the ACM manager in the form `key:value`. See the ReadiVoice SDK for information about developing ACM applications that can write to or read from this table.

Table B-5 ACM Data (cdr_acm_data)

Field Name	Field Type	Description
acm_index	SERIAL	Record index. Available for use as a water mark for processing.
conf_id	INTEGER	Conference ID of conference from which this ACM data was collected (indexed alone and with part_id).
part_id	INTEGER	Participant ID of the participant associated with this ACM data, if any.
start_time	INTEGER	Start timestamp in UNIX seconds when the ACM data was collected.
acm_name	CHAR(50)	Name of ACM data. This is the `key` sent by the ACM application.
acm_value	CHAR(50)	Collected ACM data. This is the `value` sent by the ACM application.Can be blank if no data was collected.

C

SNMP Events and Alarms

This appendix contains reference information for ReadiVoice system and bridge events and their SNMP variables. It also describes SNMP logging, which produces log files of usage and performance data, and the ReadiVoice Monitoring Tool, which provides remote (pager or email) notification of alarm events.

SNMP Reference

The ReadiVoice system supports the Simple Network Management Protocol (SNMP). It provides two built-in methods of accessing SNMP data about the system:

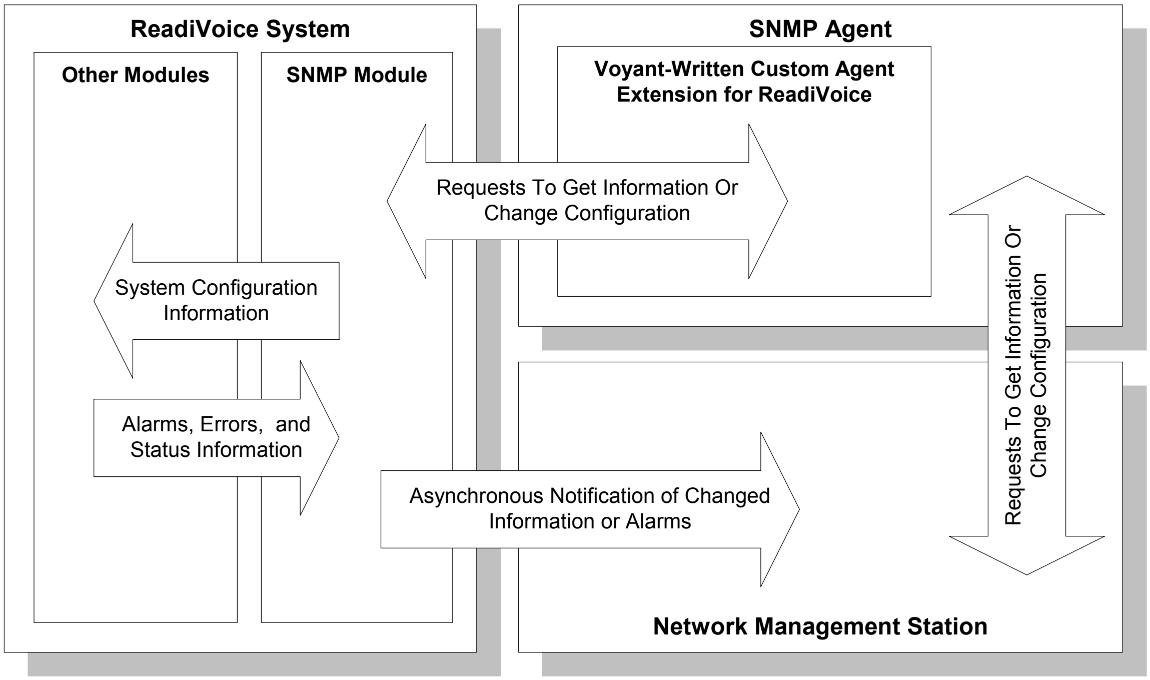
- The SNMP Monitor applet lets you monitor the system in real time, viewing data from the SNMP Management Information Base (MIB). See "Using the SNMP Monitor" on page 81.
- The SNMP logging function lets you send system usage and performance data to log files on an ongoing basis. To enable SNMP logging, see "Using the SNMP Log Files" on page 266.

To do more with SNMP, including making configuration changes, you can use standard network administration tools, such as HP Openview from Hewlett Packard or similar tools from companies such as Sun Microsystems or IBM.

Figure C-1 provides an overview of how such a network management tool interfaces with the ReadiVoice SNMP module and SNMP agent. The tables in the following section show the fields in the SNMP MIB tables.

By default, all SNMP variables are the standard SNMP data type `UInteger32` (range of values: 0…4294967295).

Figure C-1 *SNMP overview*



Contents of the SNMP MIB

The MIB contains four tables:

`sysTable` — system information

`sysHistogramTable` — system histogram (statistical) information

`brgTable` — bridge information

`brgHistogramTable` — bridge histogram (statistical) information

The tables that follow list all the fields in each MIB table.

Table C-1 *The SNMP MIB's system information table*

Field	Description
`sysNumBridgesActive`	Number of active bridges.
`sysNumModeratorDialOuts`	Number of dial-outs made by moderator.
`sysNumOperatorDialOuts`	Number of dial-outs made by operator.
`sysNumDTMFDialOuts`	Number of dial-outs made by DTMF.
`sysNumDialsIn`	Number of dial-ins.
`sysNumJoinFullFails`	Number of attempts to join full conferences.
`sysAlarmStatus`	System alarm: 0 = false; 1 = true.
`sysNumConfsActive`	Number of active conferences.
`sysModeratorUsedPrct`	Percent of conferences using moderator.
`sysPortMaxCapacity`	Total number of available ports in the system.
`sysNumPortsReserved`	Number of reserved ports in the system.
`sysNumPortsActive`	Number of active ports in the system.
`sysNumOperatorsActive`	Number of operators logged into the system.
`sysNumModeratorsActive`	Number of moderators logged into the system.
`sysOperQueueLength`	Length of operator queue.
`sysNumOperatorsPortsReserved`	Number of ports reserved for operator voice paths in the system.
`sysNumOperatorsPortsActive`	Number of operators with voice path established in the system.
`sysCrTimeout`	SS7 routing failures – timeout.

Table C-1 *The SNMP MIB's system information table (continued)*

Field	Description
`sysCrNoCapacity`	SS7 routing failures – no capacity.
`sysCrNoDBEntry`	SS7 routing failures – no DB entry (bad number).
`sysCrOutOfTranslationNum`	SS7 routing failures – out of translation number.

Table C-2 *The SNMP MIB's system histogram table*

Field	Description
`sysTime`	Hour-based time (index into the table).
`sysConfAvgSize`	Average size of the conferences for the system.
`sysMaxConfActive`	Maximum number of the conferences for the system.
`sysMaxPortsReserved`	Maximum number of reserved ports for the system.
`sysMaxPortsActive`	Maximum number of active ports for the system.
`sysOperQueueAvgWait`	Average wait time in the operator queue.
`sysOperQueueMaxWait`	Maximum wait time in the operator queue.

Table C-3 *The SNMP MIB's bridge information table*

Field	Description
`brgID`	Bridge identifier (index into the table).

`brgStatus`	Bridge status:
	`0 = unknown`
	`1 = bridgeUp`
	`2 = bridgeLogout`
	`3 = bridgeAlarm`
	`4 = bridgeNew`
	`5 = bridgeDeleted`

`brgMaintenanceStatus`	Bridge maintenance status:
	`0 = unknown`
	`1 = bridgeBusyOut`
	`2 = bridgeInService`

`brgNumModeratorDialOuts`	Number of dial outs made by moderator.
`brgNumOperatorDialOuts`	Number of dial outs made by operator.

Table C-3 The SNMP MIB's bridge information table (continued)

Field	Description
`brgNumDTMFDialOuts`	Number of dial outs made by DTMF.
`brgNumDialsIn`	Number of dial ins.
`brgNumConfsActive`	Number of active conferences.
`brgPortMaxCapacity`	Total number of available ports in the bridge.
`brgNumPortsReserved`	Number of reserved ports in the bridge.
`brgNumPortsActive`	Number of active ports in the bridge.
`brgNumOperatorsActive`	Number of active operators in the system.
`brgOperQueueLength`	Length of operator queue.
`brgOperQueueWait`	Wait time in the operator queue.
`brgOperPortsReserved`	Number of operator ports reserved on this bridge.
`brgOperPortsActive`	Number of operator voice paths established on this bridge.
`brgIpAddress`	Bridge IP address.
`brgFanAlarm`	Fan alarm: 0 = false; 1 = true.
`brgPowerSupplyAlarm`	Power supply alarm: 0 = false; 1 = true.

Table C-4 The SNMP MIB's bridge histogram table

Field	Description
`brgId`	Bridge identifier (index into the table).
`brgTime`	Hour-based time (index into the table).
`brgConfAvgSize`	Average size of the conferences for the bridge.
`brgMaxConfActive`	Maximum number of the conferences for the bridge.
`brgMaxPortsReserved`	Maximum number of reserved ports for the bridge.
`brgMaxPortsActive`	Maximum number of active ports for the bridge.

Table C-4 *The SNMP MIB's bridge histogram table (continued)*

Field	Description
`brgOperQueueAvgWait`	Average wait time in the operator queue.
`brgOperQueueMaxWait`	Maximum wait time in the operator queue.
`cfgTimeFormat`	Time format: 0 = 12-hour clock; 1 = 24-hour clock.

Using the SNMP Log Files

If you enable the SNMP logging function (see "Enabling SNMP Logging" on page 141), it writes system usage and performance data to log files on an ongoing basis. This data can be useful for trend analysis and capacity planning.

SNMP logging creates two log files:

- The SNMP log file contains selected bridge and system data.
- The histogram log file contains system usage data on an hour-by-hour basis.

Both log files are comma-separated-value (CSV) text files. The first line of each file contains the field names. You can import the files into a spreadsheet or other application for manipulation and analysis.

Every night, the SNMP logging function closes the current SNMP log file, archives it as *`filename_archivedate`*`.gz`, and begins a new SNMP log file. Once a week on Sunday night, it closes and archives the current histogram log file (naming the archive in the same way) and begins a new one.

Contents of SNMP Log File

Figure C-2 shows a sample of an SNMP log file for a three-bridge system. Because it's impossible to fit an entire line (row) from the file on one line, continuations of a single line are shown indented.

The first line in the sample contains the field names. The second and third lines contain two example records. Note that their time stamps are ten minutes apart, the polling interval used.

Table C-5 lists the fields in the SNMP log file records, along with brief descriptions and the data from the two example records shown in the sample. Note that, after the system data fields, the bridge data fields are repeated for each additional bridge in the system. In the three-bridge example shown here, these fields repeat twice (for bridges 2 and 3).

Figure C-2 *Sample SNMP log file*

```
DATE/TIME,BRIDGEID,CONFERENCES,MAX_CAPACITY,ACTIVE_PORTS,ALLOCATED_PORTS,DIAL_INS,DTMF_DIAL_
   OUTS,MODERATOR_DIAL_OUTS,OPERATOR_DIAL_OUTS,OPERATORS,OP_QUEUE,SYS_CONFERENCE,SYS_MAX_
   CAPACITY,SYS_ACTIVE_PORTS,SYS_ALLOCATATED_PORTS,SYS_DIAL_INS,SYS_DTMF_DIAL_OUTS,SYS_
   MODERATOR_DIAL_OUTS,SYS_OPERATOR_DIAL_OUTS,SYS_MODERATOR,SYS_MODERATOR_USAGE,SYS_OPERATORS,
   SYS_OP_QUEUE_LENGTH,SYS_TIMEOUTS,SYS_NO_CAPACITY,SYS_NO_DATABASE_ENTRY,SYS_OUT_OF_
   TRANSLATIONS_NUMBERS,BRIDGEID,CONFERENCES,MAX_CAPACITY,ACTIVE_PORTS,ALLOCATED_PORTS,DIAL_
   INS,DTMF_DIAL_OUTS,MODERATOR_DIAL_OUTS,OPERATOR_DIAL_OUTS,OPERATORS,OP_QUEUE,BRIDGEID,
   CONFERENCES,MAX_CAPACITY,ACTIVE_PORTS,ALLOCATED_PORTS,DIAL_INS,DTMF_DIAL_OUTS,MODERATOR_
   DIAL_OUTS,OPERATOR_DIAL_OUTS,OPERATORS,OP_QUEUE
Tue May 2 16:30:11 2000,1,1,0,480,0,0,0,0,0,0,0,0,3,1440,12,20,12,0,0,0,0,0,0,0,0,
   0,0,0,2,1,3,480,12,20,12,0,0,0,0,0,3,1,0,480,0,0,0,0,0,0,0,0
Tue May 2 16:40:12 2000,1,1,2,480,4,8,2,1,1,0,0,0,5,1440,16,22,14,1,1,0,2,40,0,0,0,
   0,0,0,2,1,3,480,12,14,12,0,0,0,0,0,3,1,0,480,0,0,0,0,0,0,0,0
```

Table C-5 *SNMP log file fields*

Field	Description	Example 1	Example 2
DATE/TIME	Date/time stamp of record (local time on CACS server)	Tue May 2 16:30:11 2000	Tue May 2 16:40:12 2000
BRIDGEID	Bridge ID	1	1
BRIDGE_STATUS	Bridge status, either 0 (off line) or 1 (on line)	1	1
CONFERENCES	Number of active conferences for the bridge	0	2
MAX_CAPACITY	Port capacity available	480	480
ACTIVE_PORTS	Ports in use – active for the bridge	0	4
ALLOCATED_PORTS	Ports reserved for the bridge	0	8
DIAL_INS	Number of dial-ins for the bridge	0	2
DTMF_DIAL_OUTS	DTMF dial-outs for the bridge	0	1
MODERATOR_DIAL_OUTS	Moderator (GUI) dial-outs for the bridge	0	1
OPERATOR_DIAL_OUTS	Operator (GUI) dial-outs for the bridge	0	0
OPERATORS	Number of operators for the bridge	1	1
OP_QUEUE	Operator queue length (number) for the bridge	0	0

Table C-5 *SNMP log file fields (continued)*

Field	Description	Example 1	Example 2
SYS_CONFERENCE	Number of active conferences for the system	3	5
SYS_MAX_CAPACITY	Port capacity available (number of ports in system)	1440	1440
SYS_ACTIVE_PORTS	Ports in use – active across the system	12	16
SYS_ALLOCATATED_PORTS	Ports in use – reserved across the system	20	22
SYS_DIAL_INS	Number of dial-ins for the system	12	14
SYS_DTMF_DIAL_OUTS	Number of dial-outs for the system	0	1
SYS_MODERATOR_DIAL_OUTS	Number of moderator dial-outs for the system	0	1
SYS_OPERATOR_DIAL_OUTS	Number of operator dial-outs for the system	0	0
SYS_MODERATOR	Number of moderators on the system	0	2
SYS_MODERATOR_USAGE	Percentage of conferences using the moderator GUI	0	40
SYS_OPERATORS	Number of operators for the system	1	1
SYS_OP_QUEUE_LENGTH	Operator queue length for the system	0	0
SYS_TIMEOUTS	SS7 routing failures – timeouts	0	0
SYS_NO_CAPACITY	SS7 routing failures – no capacity	0	0
SYS_NO_DATABASE_ENTRY	SS7 routing failures – no DB entry	0	0
SYS_OUT_OF_TRANSLATIONS_NUMBERS	SS7 routing failures – out of translation numbers	0	0
BRIDGEID	Bridge ID	2	2
BRIDGE_STATUS	Bridge status, either 0 (off line) or 1 (on line)	1	1
CONFERENCES	Number of active conferences for the bridge	3	3

Table C-5 *SNMP log file fields (continued)*

Field	Description	Example 1	Example 2
MAX_CAPACITY	Port capacity available	480	480
ACTIVE_PORTS	Ports in use – active for the bridge	12	12
ALLOCATED_PORTS	Ports in use – reserved for the bridge	20	14
DIAL_INS	Number of dial-ins for the bridge	12	12
DTMF_DIAL_OUTS	DTMF dial-outs for the bridge	0	0
MODERATOR_DIAL_OUTS	Moderator (GUI) dial-outs for the bridge	0	0
OPERATOR_DIAL_OUTS	Operator (GUI) dial-outs for the bridge	0	0
OPERATORS	Number of operators for the bridge	1	1
OP_QUEUE	Operator queue length (number) for the bridge	0	0
BRIDGEID	Bridge ID	3	3
BRIDGE_STATUS	Bridge status, either 0 (off line) or 1 (on line)	1	1
CONFERENCES	Number of active conferences for the bridge	03	0
MAX_CAPACITY	Port capacity available	480	480
ACTIVE_PORTS	Ports in use – active for the bridge	0	0
ALLOCATED_PORTS	Ports in use – reserved for the bridge	0	0
DIAL_INS	Number of dial-ins for the bridge	0	0
DTMF_DIAL_OUTS	DTMF dial-outs for the bridge	0	0
MODERATOR_DIAL_OUTS	Moderator (GUI) dial-outs for the bridge	0	0

Table C-5 *SNMP log file fields (continued)*

Field	Description	Example 1	Example 2
OPERATOR_DIAL_OUTS	Operator (GUI) dial-outs for the bridge	0	0
OPERATORS	Number of operators for the bridge	1	1
OP_QUEUE	Operator queue length (number) for the bridge	0	0

Contents of Histogram Log File

Figure C-3 shows a sample of a histogram log file. The first line in the sample contains the field names. The remaining lines are example records. The logging function writes a record to the histogram log every hour on the hour.

Table C-6 lists the fields in the histogram log records and describes them briefly. Note that the data is for the entire system (all bridges) and is a snapshot as of the time stamp, not an average for the one-hour period.

Figure C-3 *Sample histogram log file*

```
DATE/TIME,SYS_MAX_PORTS_RESERVED,SYS_CONF_AVG_SIZE,SYS_MAX_CONF_ACTIVE,SYS_MAX_PORTS_ACTIVE
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
06/05/00 0:00,30,5,5,24
```

Table C-6 *Histogram log file fields*

Field	Description
DATE/TIME	Date/time stamp of record (local time on CACS server)
SYS_MAX_PORTS_RESERVED	Number ports reserved for the entire system
SYS_CONF_AVG_SIZE	Average size (number of ports) of all currently active conferences
SYS_MAX_CONF_ACTIVE	Number of currently active conferences
SYS_MAX_PORTS_ACTIVE	Number of currently active conferences ports

D

ReadiVoice-IP with IP Tributaries

ReadiVoice-IP systems can be configured to enable the IP Tributaries feature. This feature provides alternative access methods for initiating and entering a conference. These methods perform authentication and call routing, and thus can bypass elements of the front-end call flow.

To use this feature, you must provide an authentication mechanism separate from the ReadiVoice system. The authentication mechanism puts one or more pieces of the caller's identifying information into the SIP `INVITE` message used to access the system.

What are IP Tributaries?

IP tributaries are defined as part of the `TO` header within the SIP `INVITE` message sent to the ReadiVoice CACS. The following sections describe the content of this `TO` header.

Single Number, Traditional Access

In the ReadiVoice single number, traditional access call flow, the SIP `INVITE` message has a `TO` header in the following format:

```
TO: sip:accessnumber@host
```

When the header has this format, the caller is routed to the ReadiVoice IVR system and asked to enter identifying information. The caller who keys in this information is then routed to the conference and enters the traditional ReadiVoice call flow.

Access Code Included

In this type of SIP `INVITE`, the `TO` header includes the access code. It has the following format:

```
TO: sip:accessnumber@host;AC=accesscode
```

When the header has this format, the caller is routed directly to the conference (rather than being prompted for an access code) and enters the traditional ReadiVoice call flow.

In this case, the access code has already been validated.

Password Included — One-Password Call Flow

ReadiVoice-IP has three possible IP tributaries for one-password call flows.

Clear Text Password

In this type of SIP `INVITE`, the `TO` header includes the access code and password. It has the following format:

```
TO: sip:accessnumber@host;AC=accesscode;PCODE=password
```

When the header has this format, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for an access code and password). If the password within the header is invalid, the system sends a `401 Unauthorized` response to the `INVITE`.

`PCODE=""` is valid if the participant password field in the database is empty.

Per Call Authorization

In this type of SIP `INVITE`, the `TO` header includes the access code and verification of authorization. It has the following format:

```
TO: sip:accessnumber@host;AC=accesscode;AUTH=TRUE
```

When the header has this format and the SIP protocol authorization is valid, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for an access code and password). If the authorization within the header is invalid, the system sends a `401 Unauthorized` response to the `INVITE`.

Trusted Proxy

In this type of SIP `INVITE`, the `TO` header includes the access code and participant type. It has one of the following formats:

```
TO: sip:accessnumber@host;AC=acccesscode;PTYPE=SUBSCRIBER
```

or

```
TO: sip:accessnumber@host;AC=acccesscode;PTYPE=PARTICIPANT
```

When the header has one of these formats and the `INVITE` comes from a trusted proxy, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for an access code and password). If the proxy is not a trusted proxy, the system sends a `401 Unauthorized` response to the `INVITE`. If the `PTYPE` specified is not valid, the system sends a `400 Bad Request` response to the `INVITE`.

Password Included — Two-Password Call Flow

ReadiVoice-IP has three possible IP tributaries for two-password call flows.

Clear Text Password

In this type of SIP `INVITE`, the `TO` header includes the password, but the access code is not included or is empty. It has one of the following formats:

```
TO: sip:accessnumber@host;AC="";PCODE=password
```

or

`TO: sip:`*`accessnumber@host`*`;PCODE=`*`password`*

When the header has one of these formats, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for a password). If the password within the header is invalid, the system sends a `401 Unauthorized` response to the `INVITE`.

`PCODE=""` is also valid if the participant password field in the database is empty.

Per Call Authorization

In this type of SIP `INVITE`, the `TO` header includes the verification of authorization, but the access code is not included or is empty. It has one of the following formats:

`TO: sip:`*`accessnumber@host`*`;AC="";AUTH=TRUE`

or

`TO: sip:`*`accessnumber@host`*`;AUTH=TRUE`

When the header has one of these formats, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for a password). If the authorization within the header is invalid, the system sends a `401 Unauthorized` response to the `INVITE`.

Trusted Proxy

In this type of SIP INVITE, the TO header includes the participant type, but the access code is not included or is empty. It has one of the following formats:

`TO: sip:`*`accessnumber@host`*`;AC="";PTYPE=SUBSCRIBER`

or

`TO: sip:`*`accessnumber@host`*`;AC="";PTYPE=PARTICIPANT`

or

`TO: sip:`*`accessnumber@host`*`;PTYPE=SUBSCRIBER`

or

`TO: sip:`*`accessnumber@host`*`;PTYPE=PARTICIPANT`

When the header has one of these formats and the INVITE comes from a trusted proxy, the caller enters the traditional ReadiVoice call flow after the password prompt (rather than being prompted for a password). If the proxy is not a trusted proxy, the system sends a `401 Unauthorized` response to the INVITE. If the `PTYPE` specified is not valid, the system sends a `400 Bad Request` response to the INVITE.

Implementing IP Tributaries

To implement IP tributaries:

1 Voyant technical support must modify the `.suarc` configuration file and enable the IP tributary configuration parameters.

2 You must create the mechanism (email, outside IVR system, website, etc.) for collecting the correct information and formatting the URI of the INVITE sent to the CACS.

You may implement multiple IP tributaries; none of these methods are mutually exclusive.

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