Note

Before using this information and the product it supports, read the general information under "Notices" on page 609.
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About this information

This book describes how to configure IBM® WebSphere® Voice Response for AIX® with DirectTalk® Technology so that you can develop voice applications and run them in production.

Throughout this book, WebSphere Voice Response for AIX with DirectTalk technology is referred to as WebSphere Voice Response.

Who should use this information

This book is for system administrators, system operators, and, sometimes, application developers. To use this book successfully, you must be familiar with AIX for the pSeries® and with AIXwindows®. Read the WebSphere Voice Response for AIX: General Information and Planning book before using this book. You might also find it useful to read the WebSphere Voice Response for AIX: User Interface Guide book.

How to use this information

This book contains instructions and reference information. The instructions explain how to complete activities using the WebSphere Voice Response graphical user interface.

The instructions begin with introductory information, followed by step-by-step procedures. The introductory information explains when to use the instructions and contains prerequisites for using the procedures successfully. Before you use any procedure for the first time, read through all the background information.

- When you have installed WebSphere Voice Response, read Chapter 1, “Getting started,” on page 1, which tells you how to start using the system and how to give people access.
- Then follow the instructions in the appropriate chapter to make WebSphere Voice Response ready for developing and running applications.
- When you need to set parameter values, see Appendix A, “System parameters,” on page 179 and Appendix B, “System parameter templates,” on page 575. To comply with local telephone network standards, some of these parameters must be set by the IBM representative. These include the call progress tone parameters.
Following the procedures in this information

The procedures in this information assume that you are already familiar with using a mouse in a window environment and that you know how to use the common actions such as **Save** to work with information. You will find information about using the WebSphere Voice Response windows in the *WebSphere Voice Response for AIX: User Interface Guide* book.

When you become more familiar with WebSphere Voice Response, you might think of ways to combine procedures, combine parts of procedures, or execute the steps of a procedure in a different order. Many procedures include one or more sets of numbered steps that are *alternatives*. You might not have to complete all the steps in a sequence.

---

**Typographic conventions**

This book uses the following typographic conventions:

**boldface**
Identifies an **item** that is in a WebSphere Voice Response window. The item might be a keyword, an action, a field label, or a pushbutton. Whenever one of the steps in a procedure includes a word in boldface, look in the window for an item that is labeled with that word.

**boldface italics**
Are used for emphasis. *Take extra care* wherever you see bold italics.

**italics**
Identify one of the following:
- New terms that describe WebSphere Voice Response components or concepts. A term that is printed in italics is usually followed by its definition.
- Parameters for which you supply the actual names or values.
- References to other books.

**monospace**
Identifies one of the following:
- Text that you type in an AIX window. Because AIX is case sensitive, ensure that you type the uppercase and lowercase characters exactly as shown.
- Names of files and directories (path names).
Accessibility

WebSphere Voice Response for AIX is a voice application enabler. The applications that are developed to run on WebSphere Voice Response provide telephone access to business data and services. In this way, WebSphere Voice Response provides accessibility for people who cannot access the data and services by using regular Web pages or traditional graphic interfaces. These telephone user interfaces are fully accessible to people who are blind or have low vision and, if speech recognition is used, to people with mobility impairments or limited hand use. Speech recognition capability can be provided by IBM WebSphere Voice Server, or other MRCP-V1.0-compliant speech recognition products. In addition, support for users of Telephony Devices for the Deaf (TDD) is provided as part of the WebSphere Voice Response product.

With WebSphere Voice Response you can perform many application development and system administration tasks with a text editor or line commands—these are accessible if you use a screen reader product to interface with them. Also, the default settings of the WebSphere Voice Response graphical user interface can be changed to produce large fonts and high contrast colors. Details of how to use these accessibility features can be found in the WebSphere Voice Response for AIX: User Interface Guide book. Alternatively, application development can be done with Java™ or VoiceXML development tools that are supplied by IBM and third parties.

You can also use a screen-reader product to access the WebSphere Voice Response publications in HTML format (for details of their availability see “List of WebSphere Voice Response and associated documentation” on page 647).

Notes on terminology

- A glossary of commonly-used terms is at the end of this book.
- The full product name of WebSphere Voice Response for AIX with DirectTalk Technology is generally abbreviated in this book to WebSphere Voice Response.
- The term pSeries is generically used in this book to refer both to PCI-based RS/6000® computers and to appropriate models of the System p5® and pSeries ranges. (Consult your IBM representative for details of models that are supported for use with WebSphere Voice Response.) RS/6000 computers with an MCA bus are not supported.
- The IBM Quad Digital Trunk Telephony PCI Adapter is generally referred to in this book by its abbreviation DTTA. This adapter is a replacement for the IBM ARTIC960RxD Quad Digital Trunk PCI Adapter, which is generally referred to by the abbreviation DTXA. The DTXA is not supported with WebSphere Voice Response Version 6.1.
References made to the VoiceXML 2.1 specification are intended to include VoiceXML 2.0 unless otherwise specified.

Where to find more information

The information provided in the WebSphere Voice Response library will help you complete WebSphere Voice Response tasks more quickly. A complete list of the available publications and where you can obtain them is shown in "List of WebSphere Voice Response and associated documentation" on page 647.

Useful Web sites

The following Web sites are useful sources of information about WebSphere Voice Response and related products:

WebSphere Voice Response

IBM WebSphere developerWorks resources (including WebSphere Voice products)

VoiceXML Version 2.0 and 2.1 specifications
http://www.w3.org/TR/voicexml21/
http://www.w3.org/TR/voicexml20/

CCXML Version 1.0 specification
http://www.w3.org/TR/2011/PR-ccxml-20110510/

Genesys
For more information on Genesys products go to the Genesys Web site at http://www.genesyslab.com

Making comments on this book

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IBM United Kingdom Laboratories,
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Winchester, Hampshire,
SO21 2JN, United Kingdom

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Chapter 1. Getting started

This chapter describes:
- “Logging on to WebSphere Voice Response” on page 3
- “The ASCII console” on page 4
- “Giving people access to WebSphere Voice Response” on page 5
- “Introducing the system parameters” on page 12

Logging on to WebSphere Voice Response

When WebSphere Voice Response is installed, you must start it and log on to perform the tasks described in this book. This section assumes that you are using a terminal or workstation that is running in an X-Windows environment. If you do not have a graphical display, see “The ASCII console” on page 4.

For information about using WebSphere Voice Response from a remote terminal, see the WebSphere Voice Response for AIX: User Interface Guide book.

Prerequisites
- You need to know the name of your display. This is normally stored in the $DISPLAY variable. To find the value of the variable, type the following command on an AIX command line and press Enter:
  ```bash
echo $DISPLAY
  ```
  If this variable has not been set, contact your system administrator.
- You need to know the name and password of the AIX account that has been set up for WebSphere Voice Response. The default account is dtuser.

Procedure

If WebSphere Voice Response is not already running, you must start it before logging on. If WebSphere Voice Response is running, you can log on using the Access menu (see Logging on when WebSphere Voice Response is already running).

1. Starting WebSphere Voice Response: Log on to the AIX user account that is set up for WebSphere Voice Response (normally dtuser).

2. If you are using the Common Desktop Environment (CDE):
   a. Start a terminal session.
   b. Type the following command, then press Enter:
      ```bash
      vaenit
      ```
      The system displays the Login menu:
c. Go to Step 3.

3. If you are not using the Common Desktop Environment, and the account is set up correctly, the system displays the Login menu:

If the Login menu is not displayed, type the following command on the AIX command line, then press Enter:

```
vaemin
```

4. At the Login menu, type 1, then press Enter.
   If you type 2, you can get back to this point by typing the vaemin command.

5. If $DISPLAY has not been set, you are prompted for your display name. If you are prompted, type your display name in the following format:

   `name: number`

   where `name` is the name of the display, and `number` is the session number (normally 0); for example, magpie:0. Press Enter.

6. The system displays the Status window; most of the time, you can keep this window minimized.
   The system then displays the Welcome window and the Logon window:
If you have not yet created any other administrator profiles, type `admin` in the **Administrator Profile Name** field; otherwise, type the name of your administrator profile. Note that the field is case-sensitive, so be careful to type uppercase and lowercase characters, as appropriate.

7. Type your password in the **Password** field. This field is also case-sensitive. The supplied password for `admin` is also `admin`.

   **Attention:** It is strongly recommended that you change this password immediately; for more information, see "Changing an administrator password" on page 12.

8. Click **Logon**, or press Enter.

   The system activates all menus on the Welcome window menu bar:

Logging on when WebSphere Voice Response is already running:

From the Welcome window, select **Access —> Logon**.

The system displays the Logon window. Follow Steps 6 on page 2 to 8.
You are logged on and ready to start using WebSphere Voice Response. Most of the tasks involved in configuring the system are available from the **Configuration** menu. You will probably also have to use the **Operations** menu.

When you have logged on, you should change your password to prevent unauthorized access; but first you need to know about giving access to yourself and other people (see "Giving people access to WebSphere Voice Response" on page 5).

### The ASCII console

If you do not have a graphical display, you can still perform many of the tasks by using the **ASCII console**. This is a text-only front end for the WebSphere Voice Response software. Because it does not depend on graphics, you can start it from any terminal that AIX supports. For example, this could be a terminal connected to the pSeries computer through the RS-232 (serial) interface either directly, or via a modem link.

To start the ASCII console, you need a WebSphere Voice Response administrator profile with access to the Configuration and Operations functions. If you already have an administrator profile, you do not need a new one.

If you log in remotely to use the ASCII console, you must know both the WebSphere Voice Response AIX account ID (normally `dtuser`) and password, and the WebSphere Voice Response administrator profile and password.

Ensure WebSphere Voice Response is already running. If it is not running, start it by typing `vaehinit` on the command line of a graphical display, or run `vaehinit.nox` (no Xwindows) at a separate login prompt.

To start the ASCII console:
1. Log into AIX as **dtuser**.
2. Activate an AIXterm or login window.
3. `cd` to `$VAEBIN`.
4. In the AIXterm or login window, type
   ```
   AC
   ```
   and press Enter.
5. At the prompt, type your WebSphere Voice Response administrator name and press Enter.
6. At the prompt, type your WebSphere Voice Response administrator password and press Enter.
The system displays the ASCII Console Main Menu.

For more information about using the ASCII console, see the WebSphere Voice Response for AIX: User Interface Guide book.

---

**Giving people access to WebSphere Voice Response**

To use the WebSphere Voice Response windows, you must log on to the AIX user account that has been set up for the purpose. Then log on to WebSphere Voice Response using the name of an *administrator profile* and the password that is associated with that profile. The AIX user account is normally `dtuser`; for more information about this, refer to the WebSphere Voice Response for AIX: Installation book.

**Administrator profiles**

An administrator profile controls access to the menu options and specifies the national language that is to be used for window text (if more than one language has been defined). Each user of the WebSphere Voice Response windows must know the name and password of an administrator profile, although they might not be performing administration tasks. Callers who use voice applications (end users) do not need a profile.

**How many people can use an administrator profile?**

More than one person can log on using the same administrator profile at the same time. For example, with one administrator profile, one system console, and two Xstations, three people can log on to WebSphere Voice Response at the same time.

**What administrator profiles are supplied?**

When installed, the system includes three administrator profiles whose names are `admin`, `field`, and `lab`. The `admin` profile is a starter profile, whose password is initially set to "admin", and which has access to all functions. The `field` and `lab` profiles are reserved for use by WebSphere Voice Response.

You can continue to use the `admin` profile (but do change the password to prevent unauthorized use). You can also create additional profiles for other users.

**Why create additional administrator profiles?**

You might want to create additional administrator profiles because:

- Different users want different debug settings and toolbar preferences
- Different users want to use the interface in different national languages
- You want to restrict access by some users to some of the menu options
Language preference

An administrator profile specifies a language preference. The language preference determines both the language in which window text displays after you log on (discussed in “Introducing window text” on page 166) and the locale.

Access control

If everyone uses the same administrator profile, everyone has the same access privileges and this might not be desirable. For example, you might prefer to set up a profile for the system administrator (giving access to all the menu options) and another for voice application developers (denying access to System Configuration, Pack Configuration, Administrator Profiles, and perhaps some other options). For more information about what each menu option does, refer to the WebSphere Voice Response for AIX: General Information and Planning book, or the online help for the Welcome Window.

When you set up an additional administrator profile, you can control access as shown in Table 1 on page 7. This table also shows how many people can use each menu option at the same time.

How many people can access WebSphere Voice Response at the same time?

Some menu options can be used by only one person at a time, as shown in Table 1 on page 7. If you select one of these options when it is already in use, although that user might be using the same administrator profile, the system displays the message “Program start request was denied” to inform you that the function is unavailable.
Table 1. Controlling access to menu options

<table>
<thead>
<tr>
<th>Menu</th>
<th>Available control over options</th>
<th>Number of people at the same time</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configuration</td>
<td>You can permit or deny access to each of the following menu options:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Pack Configuration → Change</td>
<td>One</td>
</tr>
<tr>
<td></td>
<td>- System Configuration → Change</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Many people can be authorized to use these options, but only one person at a time can actually select the Change option on either Pack Configuration or System Configuration.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Pack Configuration → Browse</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>- System Configuration → Browse</td>
<td></td>
</tr>
<tr>
<td></td>
<td>You have no control over these options; anybody who can log on can select the Browse option on these menu options.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- 3270 Session Configuration</td>
<td>One</td>
</tr>
<tr>
<td></td>
<td>- Administrator Profiles</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>- Application Profiles</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>- Subscriber Classes</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>- Languages</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>- Help Editor</td>
<td>Many</td>
</tr>
</tbody>
</table>
### Table 1. Controlling access to menu options (continued)

<table>
<thead>
<tr>
<th>Menu</th>
<th>Available control over options</th>
<th>Number of people at the same time&lt;sup&gt;1&lt;/sup&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operations</td>
<td>You can permit or deny access to the whole menu.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• 3270 Session Manager</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>• Custom Server Manager</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>• System Monitor</td>
<td>Many</td>
</tr>
<tr>
<td></td>
<td>• Immediate Shutdown</td>
<td>One</td>
</tr>
<tr>
<td></td>
<td>• Quiesce Shutdown</td>
<td>One</td>
</tr>
<tr>
<td></td>
<td>• Statistics</td>
<td>Many</td>
</tr>
<tr>
<td>Applications</td>
<td>You can permit or deny access to the whole menu.</td>
<td>Many</td>
</tr>
</tbody>
</table>

**Note:**
1. For a single system image, ‘One’ in this column means one person at each node.

The **Pack Configuration —&gt; Change** and **System Configuration —&gt; Change** options are handled as one option; access is restricted to one person at a time to one of these options (see [Introducing the system parameters](#) on page 12). However, anybody can click **Pack Configuration —&gt; Browse** or **System Configuration —&gt; Browse** at any time. More than one person at a time can access other options (see [Table 1 on page 7](#)), but the system does not tell you when someone else is also using the option; the system does not prevent two people from modifying the same information at the same time, possibly with confusing results. Therefore it is very important for everyone using the system to coordinate with each other so that two people do not change, delete, or save any of the following at the same time:

- State table
- Prompt
- Prompt directory
- Custom server
- 3270 script
- 3270 screen definition
Creating administrator profiles

Each time someone new starts using WebSphere Voice Response, consider what menu options they need to be able to use, and what their language preference is (if you have more than one language defined). If no suitable administrator profile is available, you can either create a new one from scratch, or copy an existing one and change it. To do either, you need to be logged on with a profile that gives access to the Administrator Profiles menu option. You can copy any profile, but you can open and change only a profile that has fewer access privileges than the profile you are logged on with.

For each profile, you need an administrator name and a password. Both can be up to eight characters long. You can use letters only, numbers only, or a combination of both.

Note: When you create an administrator profile on a single system image, the new profile information does not take effect on a client node until you restart WebSphere Voice Response on that node.

Creating a new administrator profile

Use this procedure to create a completely new administrator profile. (To copy an existing profile, use the procedure in "Copying an administrator profile" on page 11.)

Procedure

1. From the Welcome window, select Configuration —> Administrator Profiles.
2. Creating a new profile: Click File —> New. The system displays the Administrator Profile window:
3. **Entering the password**: Type in the password for this administrator profile. The password is displayed when you type. For security reasons, the password of an administrator profile that has been saved is never displayed again.

4. **Defining the preferred language**: Click **Language**. The system lists the languages that have been defined on your system.

5. Click the preferred language for this administrator profile and click **OK**. The system displays the Administrator Profile window with the language filled in.

6. **Defining access rights**: Click the functions that you want this administrator profile to be able to access.

   **Note**: For **System Configuration**, you can select one of the following levels: Admin, User2, User3, User4, or Field. However, the current release of WebSphere Voice Response recognizes only one level, Admin, which gives *write permission*. User2, User3, or User4 give browse permission for System Configuration and Pack Configuration, but nothing else. Field is for use only by an IBM representative.

7. **Saving the profile**: Save the profile. The system prompts you for the administrator profile name.

8. Type in the name that people will use to log on.

9. Click **OK**.
The system saves the new profile and creates the administrator profile. The Administrator Profile window displays the new name in the title bar. When you Close the window, the system lists the new profile.

10. **Creating additional administrator profiles** : To create another administrator profile, click File —> New. Repeat this procedure starting with Step 1.

11. Click Close.

**Copying an administrator profile**

Use this procedure to create a new administrator profile by copying an existing profile. To create a new profile from scratch, use the procedure in “Creating a new administrator profile” on page 9.

To copy a profile, you must be logged on with a profile that has access to the Administrator Profiles function. You can copy any profile, but you can open and change only a profile that has fewer access privileges than the profile you logged on with.

After you copy a profile, ensure that you assign a new password to it. Until you assign a new password, no user can use the copied profile to log on.

**Procedure**

1. From the Welcome window, select **Configuration —> Administrator Profiles**.

2. **Copying a profile** : Click the profile to copy.

3. Click **File —> Copy**.

   The system prompts you for the name of the new profile.

4. Type in the name of the new profile.

5. Click **OK**.
The system displays the Administrator Profiles window, which shows the new profile.

6. **Registering the password**: Open the new profile.

   The system displays the Administrator Profile window. The name at the top identifies the new profile. The access privileges and language are the same as the privileges and language in the profile that you copied.

7. Type in a password for this profile.

8. Change any other information that does not apply to this profile.

9. **Save** the new profile.

### Changing an administrator password

Follow this procedure to reset an administrator password. The password is not displayed when you open an administrator profile. If a user forgets the password, the password must be reset.

#### Procedure

1. From the Welcome window, select **Configuration —> Administrator Profiles**.

2. **Changing the password**: Open the administrator profile.

   The system displays the Administrator Profile window.

3. Type in the new password.

4. **Save** the profile.

### Introducing the system parameters

System parameters control the operation of many aspects of the WebSphere Voice Response system and of the voice applications that are running on it. Most of the configuration tasks described in this book include the setting of system parameters. However, you have to set only the system parameters whose default values are not suitable for your operation. Default values have been set to support most systems where possible.

The system parameters are divided into several groups, which are listed and described in "System parameter groups" on page 179. The system parameters are listed and explained in "System parameters reference" on page 195. All this information is also available from the **Help** button in the system parameter windows.

### Defining multiple objects

Some groups of parameters apply to objects of which there might be more than one; for example, channels, channel groups, trunk interfaces, signaling types, call progress tones, reports, and telephone keys. To make it easy to
define these objects, you can copy the whole group of parameters for one object, and paste it on to another (see “Using system parameter templates” on page 17).

**Access to system parameters**

You can access all system parameters if you use the **System Configuration** option on the **Configuration** menu in either the Welcome window or the ASCII Console. Alternatively you can use the **wvrsysconf** command to directly manipulate the system parameters. Both of these methods are described in Chapter 4, “Defining the telephony environment (System Configuration),” on page 73.

You can more easily set parameters that control telephony operations if you use the **Pack Configuration** option on the **Configuration** menu in the Welcome Window. The **wvrteleconf** command can be used as an alternative interface. Pack configuration is discussed in more detail in Chapter 3, “Defining the telephony environment using Pack Configuration or wvrteleconf,” on page 43.

To change the value of a system parameter, you must be logged on to WebSphere Voice Response, using an administrator profile with permission to **Change System Configuration** or **Change Pack Configuration**. This gives you access to those parameters listed in “System parameters reference” on page 195 as having “Admin” access. To browse or change parameters listed as having “Field” access, you must have the password to the field administrator profile. Your IBM Representative might provide this password in some conditions. Normally, these parameters do not need to be changed.

Anybody who is logged on to WebSphere Voice Response can browse the values of system parameters listed as having “Admin” access.

**Multiple access to system parameters**

More than one person can open the System Configuration or the Pack Configuration windows. They might be using the same administrator profile, or different profiles. However, only the first person to open one of these windows can select the **Change** option. Until that person closes the System Configuration or Pack Configuration window, everyone else can select the **Browse** option only.

**Note:** When you are looking at either System Configuration or Pack Configuration, you see the values that are set now, which are not necessarily the values that the system is using. You might need to take some action before new values take effect (see “System parameter groups” on page 179).

**Setting the value of a system parameter**

To change the values of system parameters, you must be logged on with an administrator profile that gives you permission to change.
Note: Any adjustments that might affect compliance with telecommunications authority regulations are to be made only by authorized personnel who are familiar with these requirements.

Procedure
1. From the Welcome window, select Configuration --> System Configuration --> Change.

   Note: If someone already has either the System Configuration or the Pack Configuration window open for changing, you can select only Browse.

   ![](System_Configuration_window.png)

2. Selecting the parameter group: Click the name of the parameter group to which the parameter belongs. The group to which each parameter belongs is listed in "System parameters reference" on page 195.

   In some groups, you can define multiple objects of the same type (see "System parameter groups" on page 179); the system displays a window from which you select the object. (For examples, see "Defining signaling types" on page 78, "Defining channel groups" on page 80, and "Defining channels" on page 84.)

   Note: In some of the groups with multiple objects, the named objects are to be used as templates. In this event, you might want to copy a named object to a numbered object (see "Using system parameter templates" on page 17). Then select the numbered object.

   If necessary, select the object.

   The system displays the existing values of all the parameters in the group, for example:
3. **Displaying the current value:** Click the parameter, then click **File** —> **Open**.

4. **Getting help:** To see an explanation of the parameter, click **Help**.
   
The online help information explains what each parameter does and gives guidance about which value to specify. This information is also available in **Appendix A, “System parameters,” on page 179**.
5. **Changing the value:** Type the new value. For some parameters, you select the value from a list of buttons:

6. Click **OK**.
7. Close the parameter group window and any other windows until the System Configuration window is displayed.
8. When the System Configuration window is displayed, click **File —> Save**.
9. Close the System Configuration window.

**When do new values take effect?**

Changes to system parameters do not take effect immediately. Table 2 tells you what to do to make the parameters in each group take effect.

To shut down WebSphere Voice Response, click **Operations —> Immediate Shutdown** or **Operations —> Quiesce Shutdown**.

To disable then enable the appropriate trunk, click **Operations —> System Monitor**, then click the required trunk and choose the appropriate command from the menu. For more information about shutting down WebSphere Voice Response and using the System Monitor, see the *WebSphere Voice Response for AIX: Managing and Monitoring the System* book.

**Table 2. Making new system parameter values take effect**

<table>
<thead>
<tr>
<th>Parameter group</th>
<th>To make new parameter values take effect...</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application Server Interface</td>
<td>Restart WebSphere Voice Response.</td>
</tr>
<tr>
<td>Call Progress Tones</td>
<td>Disable then re-enable the packs.</td>
</tr>
<tr>
<td>Channel</td>
<td>Disable then re-enable the packs to which the channels are connected.</td>
</tr>
</tbody>
</table>
Table 2. Making new system parameter values take effect (continued)

<table>
<thead>
<tr>
<th>Parameter group</th>
<th>To make new parameter values take effect...</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel Group</td>
<td>Disable then re-enable the packs to which the channels in the channel group are connected.</td>
</tr>
<tr>
<td>CPU Monitor</td>
<td>Restart WebSphere Voice Response.</td>
</tr>
<tr>
<td>General</td>
<td>Restart WebSphere Voice Response.</td>
</tr>
<tr>
<td>Exchange Data Link</td>
<td>Restart WebSphere Voice Response.</td>
</tr>
<tr>
<td>ISDN Signaling</td>
<td>Stop the ISDN signaling process and disable all ISDN trunks, then restart the signaling process and re-enable the trunks.</td>
</tr>
<tr>
<td>Key Signals</td>
<td>Restart WebSphere Voice Response.</td>
</tr>
<tr>
<td>Signaling Type</td>
<td>Disable then re-enable the packs to which channels that use the signaling group are attached.</td>
</tr>
<tr>
<td>Trunk Interface</td>
<td>Disable then re-enable the pack.</td>
</tr>
</tbody>
</table>

Browsing system parameters

To browse system parameters, click Configuration —> System Configuration or Pack Configuration —> Browse. (If the administrator profile does not give you write access, the Change option is grayed out and you cannot select it.)

When browsing, you can change the values of parameters, but you cannot save the new values. You cannot click File —> Save. The first time you click OK to exit a system parameter window, a dialog again warns you that you cannot save your changes.

Using system parameter templates

A template is a set of values for a group of system parameters, which you can copy and paste to define all the parameters in a group at the same time. Some templates (for call progress tones, signaling types, and trunk interfaces) are supplied with WebSphere Voice Response. The supplied templates are listed in Appendix B, “System parameter templates,” on page 575.

Do not use the supplied templates directly, because you cannot edit the values in them. (Even though at first the supplied values may seem suitable, you might find later that you need to change one or more of them.)
You can also use your own definitions as templates. For example, you can define channel group 1 then copy it and paste it onto channel group 2. You can then edit some of the values in channel group 2. For example, channel group 2 could specify a different signaling type in its Signaling Type parameter.

**Copying parameter values**

Use this procedure whenever you want to copy a set of parameter values. When you have made the copy, use the procedure that explains how to edit the parameters in that particular group to change the copy.

You can use this procedure for any of the following groups:

- Channel
- Channel Group
- Signaling Type
- Trunk Interface
- Call Progress Tones (if you are logged on as Field)

**Procedure**

1. At the Welcome window, select **Configuration —> System Configuration —> Change**.
2. **Opening the parameter group**: Click the group you want.
3. **Copying a set of parameter values**: Use the arrow keys to select the template you want to copy (for example, **E1 United Kingdom** in the **Trunk Interface** group).
4. Click **Edit —> Copy**.
5. Use the arrow keys to select the target item (for example, **1** in the **Trunk Interface** group).
6. Click **Edit —> Paste**.

The values of the parameters in the target are reset to the values contained in the template.
If you need to change some of the values, use the procedure in “Setting the value of a system parameter” on page 13.

Making a backup copy of system parameter values

The system parameter values are stored in a single file. When you have reset the system parameter values (either by using System Configuration or by using Pack Configuration) and have created a satisfactory system, you should make a backup copy. Then, if something happens to the parameter file, you will not have to reset the parameter values again; instead, you can copy the backup file.

1. **Start:** Log on to AIX as dtuser.

2. **Ensuring that the environment is set correctly:** To check whether you are logged on as the WebSphere Voice Response user, open an AIX window.

3. Type the following command and press Enter:
   ```
   echo $VAE
   ```
   The system should display:
   ```
   /usr/lpp/dirTalk
   ```

4. If you are not logged on as the WebSphere Voice Response user, type the following command and press Enter:
   ```
   . /usr/lpp/dirTalk/tools/vae.setenv
   ```
   Ensure that you leave a space between the period and the backslash before `usr`.

5. **Copying the file:** Type the following command and press <Enter>:
   ```
   cp $SYSPARM_DIR/rd.data $SYSPARM_DIR/rd.data.bak
   ```
   where `rd.data.bak` is a filename of your choice.

You now have a backup copy of the system parameter file.
20 Configuring the System
Chapter 2. The telephony environment

When first installed, WebSphere Voice Response can process signals on 12 telephone channels using default trunk protocols and signals. However, WebSphere Voice Response can handle a variety of telephony equipment and signaling protocols as used in different countries and at different sites. So you can configure the system to suit your needs.

This chapter contains the following sections:

- “Overview of the telephony configuration process”
- “Telephony concepts” on page 22
- “Answering each call with an appropriate application” on page 30.

Overview of the telephony configuration process

In particular, you must decide whether you are going to have different applications to answer different calls, or whether all calls are to be answered by the same application.

1. Read the information in “Telephony concepts” on page 22 and “Answering each call with an appropriate application” on page 30.

2. Decide which telephony functions your applications are going to need, and which signaling protocols can provide these functions. Use the information that you recorded in the planning checklist in the WebSphere Voice Response for AIX: General Information and Planning book.

3. Use the Pack Configuration option to set the country or region name, to assign channels to trunks, and to define the telephony characteristics that your applications need. Follow the instructions that are given in Chapter 3, “Defining the telephony environment using Pack Configuration or wvrteleconf,” on page 43.

4. If you have an exchange data link, or are using ISDN or Signaling System Number 7, you might need to set some other system parameter values. For instructions, see Chapter 5, “Exchange data links and common channel signaling,” on page 91.

5. For most conditions, your telephony configuration is now complete. If your configuration needs additional parameter settings, use the instructions given in Chapter 6, “Advanced system parameter settings,” on page 101.
Attention

Note that any adjustments that might affect compliance with telecommunications authority requirements are to be made only by *authorized personnel* familiar with these requirements. For this reason, you must ensure that you restrict access to Configuration, especially Pack Configuration, by defining suitable administrator profiles.

When you have configured WebSphere Voice Response, stop then start WebSphere Voice Response to make the new configuration take effect. See “When do new values take effect?” on page 16 for more information.

When you have configured the telephony environment, save a copy of the system parameter values (see “Making a backup copy of system parameter values” on page 19) then activate the trunks and channels so that WebSphere Voice Response can start processing calls. The *WebSphere Voice Response for AIX: Managing and Monitoring the System* book describes how to do this.

**Telephony concepts**

WebSphere Voice Response supports either a T1 or an E1 interface with the telephone switch. With country-specific regulations, the characteristics of the switch and the signaling protocols it uses make up the *telephony environment*. The telephony environment is defined to WebSphere Voice Response by a large number of *system parameters*. Some features of switches and protocols are optional, and your configuration must specify the features that your voice applications require. Before you start to configure your telephony environment, think about what your application needs.

**The switch**

The switch is the telephone exchange to which WebSphere Voice Response is connected. Types of switch include a central office (CO) switch, a private automatic branch exchange (PABX), an automatic call distributor (ACD), or a host-controlled digital switch.

**Trunks**

A trunk is an E1 connection with up to 30 channels, or a T1 connection with up to 24 channels. Each WebSphere Voice Response system can potentially1 support up to 480 E1 channels or 384 T1 channels; that is 16 E1 trunks or 16 T1 trunks.

To specify whether E1 or T1 trunks are to be used, set the Trunk Interface parameter in the WebSphere Voice Response parameter group. Set this

---

1. The number of channels supported depends on the model of pSeries computer and the type of application. See the *WebSphere Voice Response for AIX: General Information and Planning* book for more information.
parameter for all trunks. The voice encoding scheme that is used on T1 trunks is normally μ-law. The voice encoding scheme used on E1 trunks is A-law.

The characteristics of each trunk are defined by the parameters in the Trunk Interface group.

Channels
The channels are the lines on which the voice signals are sent backward and forward between the caller and WebSphere Voice Response. Each incoming telephone call arrives on a channel, and each outgoing call is made on a channel.

Signaling protocols
In addition to the voice signals that are sent to and from the caller, other signaling information is also sent to and from the switch, to enable such functions as:

- Call control (call setup and clearing)
- Detection and notification of caller hangup
- Provision of call information such as the caller's telephone number
- Call-transfer and message-waiting information

A variety of signaling protocols provide some or all of these functions. Functions that are supported by T1 protocols are listed in Table 3 and functions that are supported by E1 protocols are listed in Table 4 on page 25. Before deciding on or configuring a specific protocol, you must ensure that your central office switch, PSTN, PABX, or channel bank can also support the required functions. More information about the protocols is given in the WebSphere Voice Response for AIX: General Information and Planning book.

Table 3. Functions provided by T1 protocols

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Type</th>
<th>Connectivity</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>PSTN</td>
<td>PABX</td>
</tr>
<tr>
<td>E&amp;M(^1)</td>
<td>Trunk</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>FXS Loop Start(^1)</td>
<td>Line</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>SAS Loop Start</td>
<td>Line</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Protocol</td>
<td>Type</td>
<td>PSTN</td>
<td>PABX</td>
</tr>
<tr>
<td>--------------------------</td>
<td>------</td>
<td>------</td>
<td>------</td>
</tr>
<tr>
<td>FXS Ground Start ³</td>
<td>Line</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>ISDN (5ESS 5E9)</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN (5ESS 5E12)</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN (T1 National 2)</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN (DMS100 BCS34/36)</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN (TR41449/41459)</td>
<td>Trunk</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>ISDN NA008 (DMS National)</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN IEC05 (DMS250)⁹</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN (INS Net Service 1500)</td>
<td>Trunk</td>
<td>No</td>
<td>Yes</td>
</tr>
</tbody>
</table>

Table 3. Functions provided by T1 protocols (continued)

Configuring the System
### Table 3. Functions provided by T1 protocols (continued)

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Type 9</th>
<th>Connectivity</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSTN</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PABX</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Channel bank</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Answer detection</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call transfer</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Far-end disconnect</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ANI</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DID or DNIS</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

In the **Type** column, Trunk means “trunk-side protocol” and Line means “line-side protocol”

**Note:**
1. For both 2-bit AB (SF) format and 4-bit ABCD (ESF) format, as defined in TIA/EIA 464-B.
2. A channel bank for 4-bit (ESF) format CAS signaling must support extended superframe (ESF) line framing.
3. Yes, if the switch offers a release link trunk.
4. Yes, if a disconnect clear signal is provided.
5. Some PABX and ACD systems send number identification by sending DTMF digits before or after the call is answered.
6. A channel bank for ISDN must support extended superframe (ESF) line framing.
7. Supports RLT call transfer and two B-channel transfer.
8. Supported RLT call transfer.

### Table 4. Functions provided by E1 protocols

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Type 12</th>
<th>Connectivity</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSTN</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PABX</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Channel bank</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Answer detection</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call transfer</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Far-end disconnect</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ANI</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DID or DNIS</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- E&M¹ Trunk Yes Yes Yes No³ Yes Yes Yes
- FXS Loop Start Line Yes Yes Yes No³ No³ No³
- EL7/CAS Line No Yes³ No Yes Yes Yes No³ No³
- Italy Trunk Yes No No Yes No Yes

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Table 4. Functions provided by E1 protocols (continued)

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Type11</th>
<th>Connectivity</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>PSTN</td>
<td>PABX</td>
<td>Channel bank</td>
</tr>
<tr>
<td>R27</td>
<td>Trunk</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>R2MFC3.9</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>RE</td>
<td>Line</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>SL10</td>
<td>Line</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>TS00311</td>
<td>Trunk</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>UK Callstream</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>UK Exchange</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>UK Tie/DDI</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>ISDN</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>SS7 ISUP</td>
<td>Trunk</td>
<td>Yes</td>
<td>No</td>
</tr>
</tbody>
</table>
Table 4. Functions provided by E1 protocols (continued)

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Type</th>
<th>Connectivity</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSTN</td>
<td></td>
<td>Channel bank</td>
<td>Answer detection</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Call transfer</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Far-end disconnect</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>ANI</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>DID or DNIS</td>
</tr>
</tbody>
</table>

In the Type column, Trunk means “trunk-side protocol” and Line means “line-side protocol”

Note:
1. For connection to Siemens Hicom 300 switch.
2. Yes, if the switch offers a release link trunk or an ACL exchange data link is available.
3. Yes, if a disconnect clear signal is provided.
4. Some PABX and ACD systems send number identification by sending DTMF digits before or after the call is answered.
5. Unique protocol for Ericsson MD110.
6. Yes, if a VMS exchange data link is available on the MD110.
7. R2 digital line signaling as specified by ITU-T Q.421.
8. Korean only
9. R2MFC uses R2 for line signaling
10. Subscriber Loop, used in France.
11. Multifrequency compelled (MFC) is not supported.
12. Yes, to support “inverted” E&M.
13. “Mid-call diversion” is supported.

Two classes of signaling protocol exist: channel associated signaling (CAS) and common channel signaling (CCS). In addition, an exchange data link (EDL) can be used with a CAS protocol to provide additional signaling functions.

**Channel associated signaling**

When channel associated signaling protocols, such as E&M wink start, or FXS loop start, carry the signaling information for a particular channel, they carry the information on either of the following:

- The voice channel.
- A signaling channel that is permanently associated with the voice channel.

This signaling information enables call control (call setup and clearing), but other functions, such as far-end hang up detection, are not necessarily...
provided by all channel associated signaling protocols.

Exchange data link

If your CAS protocol does not provide all the signaling functions that you need, you might be able to use an exchange data link; that is, a link from the switch to the pSeries computer on which WebSphere Voice Response runs. This link can carry additional signaling information to enable:

- Detection and notification of caller hangup (far-end hangup)
- Provision of call information, such as called and calling telephone numbers
- PABX-like functions, such as call transfer and message waiting indication

Exchange data link signaling protocols, such as ACL, SMDI, SMSI, and VMS, are typically used with trunks that use a channel associated signaling protocol for call control.

Alternatively, you can use CallPath Server to send additional information to and from the switch, including called and calling number, message waiting indication (MWI), and far-end hangup detection. The CallPath_SigProc signaling process that is used to communicate with CallPath Server is registered as an exchange data link, and needs to be configured in a similar way.

Note: Some switches restrict use of CallPath to trunks that use line-side protocols instead of trunk-side protocols (as shown in Table 3 on page 23 and Table 4 on page 25). Ask your switch manufacturer to verify that CallPath is supported.

Common channel signaling

Common channel signaling protocols, such as Signaling System Number 7 (SS7) and Primary Rate Integrated Services Digital Network (ISDN), carry all the signaling information for many voice channels over a single, dedicated signaling link. The common channel signaling link can provide:

- Call setup and clearing
- Detection and notification of caller hangup (far-end hangup)
- Provision of call information, such as the caller’s telephone number

The signaling link is usually one 64 Kbps channel on a T1 or E1 trunk. Most common channel signaling protocols are based on the OSI reference model (X.200), and work by exchanging predefined messages.
Supporting one or more protocols

Different protocols on each trunk

You can use a different protocol on each trunk, provided you do not mix E1 and T1 protocols on a single system. Some of the trunks can use CAS protocols, and others can use CCS.

Different CAS protocols on the same trunk

Two WebSphere Voice Response principles, signaling type and channel group, enable you to use more than one channel associated signaling (CAS) protocol on different channels on the same trunk. Different channels on the same trunk can therefore interact with the switch in different ways to provide varying features for different applications.

Signaling types:

A signaling type is the definition of a signaling protocol to be used by channels on one or more trunks. For example, one signaling type can define timing and signaling to be used on an E&M trunk that uses wink start. Another can define timing and signaling to be used on an E&M trunk that uses delay start. A third can define timing and signaling to be used on an FXS trunk.

Channel groups:

A channel group is a set of channels on one or more trunks, using the same signaling type. Every channel must belong to one and only one channel group. All the channels can belong to the same group.

You can create a total of 16 channel groups. Which channels belong to which group is your choice; define the members of each group by setting the Channel Group parameter for each channel.

Each channel group can use a different signaling protocol. For example, if the Trunk Interface type is T1, one channel group can use the E&M trunk protocol and another the FXS trunk protocol. You could have two channel groups using FXS, one to support loop start signaling, and another to support ground start signaling.

Are signaling type and channel groups necessary for CCS?

With common channel signaling (CCS), one signaling protocol provides signaling information for all channels on the trunk, so no signaling type definition is needed.
You can divide the channels into channel groups if you are using common channel signaling (CCS), although it probably is not necessary if you are using only one CCS protocol. In this condition, you can have all the channels in channel group 1. By default pack configuration assigns all the channels on a trunk into a single channel group except for trunks 13, 14, 15, and 16, which are all assigned to channel group 12.

Answering each call with an appropriate application

Configuring WebSphere Voice Response so that the correct application can handle each incoming call is one of your most important tasks. So it is important to understand how WebSphere Voice Response answers each incoming call and chooses the application to handle it.

How does WebSphere Voice Response answer an incoming call?

When WebSphere Voice Response receives an incoming call it uses the call's application profile ID to determine how it should be answered.

Initially, control of the call is always passed to the Incoming_Call state table. Incoming_Call issues the AnswerCall action, then the InvokeStateTable action to invoke the state table specified in the application profile.

If the application profile specifies a state table called JavaApplication and the Java and VoiceXML environment is installed and running, control of the call passes to a Java application (the JavaApplication state table should always be available so it can be invoked if the Java and VoiceXML environment is not running). For more information on configuring your system to answer incoming calls with Java applications, see WebSphere Voice Response for AIX: Deploying and Managing VoiceXML and Java Applications.

Each state table that is designed to handle incoming calls therefore requires an application profile, and WebSphere Voice Response must find the correct application profile before Incoming_Call can invoke the correct state table to handle the call.
SV22 and SV185 are set to the application profile ID.

SV129 is set to 0

SV129 is set to 1

SV129 is set to 2

Is WebSphere Voice Response configured to receive the called number?

Yes

Has a called number been received?

No

SV129 is set to 0

Yes

SV129 is set to 1

Is there an application profile to match the called number?

No

Is there an application profile to match the channel identification?

Yes

Is there an application profile to match the System Default Application Profile?

No

Yes

SV22 and SV185 are set to the application profile ID.

Incoming_Call answers the call and invokes the state table specified in the profile

Is state table name=Java Application and is Java and Voice XML Environment running?

No

Control is passed to a Java application

Yes

Control is passed to the state table specified by the application profile

The call is not answered

Figure 1. How WebSphere Voice Response finds the state table to handle an incoming call
How does WebSphere Voice Response find the application profile?

Incoming_Call assumes that an application profile specifying a state table has been found. Figure 1 on page 31 shows how WebSphere Voice Response finds the application profile. When a call comes in and the called number is available, WebSphere Voice Response searches for an application profile whose ID matches the called number. (If the called number was received on an exchange data link, the area code specified in the channel group is concatenated to the beginning of the number before the search.) If the called number is unavailable for any reason, WebSphere Voice Response searches for an application profile whose ID matches the channel identification (for details, see “Channel identification” on page 37). If no profile is found, WebSphere Voice Response looks for an application profile whose ID matches the value of the System Default Application Profile system parameter (whose supplied value is 0000000000).

If no qualifying application profile ID is found, the call is not answered.2

Table 5. Application profile examples

<table>
<thead>
<tr>
<th>Profile ID</th>
<th>State table name</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1234567</td>
<td>Accounts_Main</td>
<td>Tell callers their account details.</td>
</tr>
<tr>
<td>1238800</td>
<td>Interest_Main</td>
<td>Tell callers the current interest rates.</td>
</tr>
<tr>
<td>0000000000</td>
<td>Main</td>
<td>Offer callers a choice of services.</td>
</tr>
</tbody>
</table>

What happens when Incoming_Call answers the call?

After issuing the AnswerCall action, the Incoming_Call state table calls the state table specified in the application profile. Because it calls the state table by application profile, Incoming_Call cannot send any parameters to the state table. The following system variables are, however, available for the state table to use:

Application - Profile ID (SV22)
The ID of the application profile found by WebSphere Voice Response.

---

2 If WebSphere Voice Response is configured to go off hook before attempting to get call information (see “EDL Call Information After Off Hook” on page 305), the system plays the caller a message saying that there are technical difficulties, and then hangs up.
System : Call Info : Called Number (SV185)
The called number or, if no called number is available, the channel identification.

Caller - Profile ID (SV20)
The calling number (if available).

System : Call Info : Calling Number (SV186)
The calling number (if available).

System : Call Info : Call info status (SV129)
Indicates whether call information (called number and calling number) was received. If the value of this variable is 1 (successful), SV185 is the called number; if the value is 0 (undefined) or 2 (failed), it is the channel identification.

What happens if the state table is not valid?

If a problem occurs with the state table that is specified in the application profile, for example, if it is not found or is not valid, Incoming_Call calls another supplied state table, Welcome, which greets the caller and hangs up. Welcome is also called during the installation test that is described in the WebSphere Voice Response for AIX: Installation book: WebSphere Voice Response finds the application profile whose ID is 0000000000, and this application profile specifies Welcome as the state table.

Do Incoming_Call and Welcome need customizing?

Incoming_Call and Welcome are set up and ready to use. You should record a new voice segment for Welcome to play, perhaps explaining to the caller that the system is experiencing technical difficulties.

If your channels are controlled by a CCS signaling process such as ISDN or SS7, you can customize Incoming_Call to make use of call setup information contained in the Call Info: Info In System Variable (SV542). The information is available at the onset of the state table and is then overwritten when the AnswerCall action is issued. For more information, see WebSphere Voice Response for AIX: Application Development using State Tables.

Both of these applications are imported into WebSphere Voice Response as part of the installation process.

Do not delete the state table specified in the State Table Name for Incoming Calls parameter (Incoming_Call), because WebSphere Voice Response cannot answer calls without it. Do not delete the application profile specified in the
System Default Application Profile parameter (0000000000), because
WebSphere Voice Response might need it to answer a call if no other
application profile can be found.

If the state table or application profile are modified or deleted by mistake, you
can reimport them from /usr/lpp/dirTalk/sw/samples/BaseData.imp.

**Can you write your own state table to answer calls?**

If you prefer to write your own state table to answer incoming calls, you can.
If you choose a different name for it, you must change the State Table Name
for Incoming Calls parameter in the Application Server Interface parameter
group. After you have changed the name, stop then start WebSphere Voice
Response to make the new value take effect.

If your “state table for incoming calls” does not include an AnswerCall action,
each state table that is called by it must include its own AnswerCall action.

**Handling incoming calls in your production system**

If you do not change Incoming_Call, several methods of handling incoming
calls are still available, and your decision partly depends on the capabilities of
the switch to send the called number on to WebSphere Voice Response. You
must decide whether to:

- Have one state table to handle all incoming calls
- Choose the state table on the basis of the called number
- Dedicate a group of channels to each voice response service, and let the
  switch make the decision with reference to the called number

**Have one state table to handle all incoming calls:**

With this method, the state table asks the caller which service is required, and
calls another state table. You need only one application profile to make this
work. You can use one of the following methods:

- Make the application profile ID match the value of the System Default
  Application Profile parameter in the WebSphere Voice Response parameters
group
- Give all your channels the same identification (the same area code and
  phone number) and make the application profile ID match this
  identification.

**Choose the state table with reference to the called number:**

With this method, WebSphere Voice Response can respond with the
appropriate voice response service for each caller. Each caller dials a different
number depending on which service is required, and gets directly through to that service. In the example shown in Table 5 on page 32, callers dial 1234567 for the accounts application (Accounts_Main) or 1238800 for the interest rates application (Interest_Main).

You should have one application profile for each state table that might be called in this way. You also need a default application profile and state table. Typically, the default state table asks the caller which service is required, in the event that the called number is unavailable for some reason outside your control.

This method is the most effective of the three methods that are described in this section. However, if your switch cannot send the called number to WebSphere Voice Response, you cannot use this method.

Dedicate a group of channels to each voice response service:

This method has the same advantage as choosing the state table on the basis of called number. The disadvantage of this method is that it ties channels to specific applications, which is inefficient if all the calls at one time are for one of the applications: it makes load-balancing difficult. If it is possible to get the called number from the switch, you should use that method.

Again, you need one application profile for each state table that might be invoked in this way.

Configuring WebSphere Voice Response to get the called and calling numbers

How you get the called number depends on what information about the call is available, in your country, from the switch. These methods are available:

- Dialed number (DID or DNIS) (see “Dialed number (DID or DNIS)” on page 36)
- Signaling link (see “Common channel signaling (CCS) Link” on page 36)
- Exchange data link (see “Exchange data link” on page 36)

In addition, unless you decide to use the System Default Application Profile, you should assign a number to each channel for use when the called number is not available (for example, because it was never sent to your switch). This number is known as:

- Channel Identification (see “Channel identification” on page 37)

---

3. The called number might be unavailable if, for example, the switch did not receive the full information, or because the task the switch uses to send the information is busy.
Note: Whichever way the information arrives, it is assigned to the same system variable, System : Call Info : Called Number (SV185). This means that you can change the method of answering an incoming call without having to change your application profiles or your applications. The maximum length of the number assigned to SV185 is 20 characters, including the area code. If the number is longer than this, it is truncated and an error is logged.

Dialled number (DID or DNIS)

The dialled number can be provided by the dialled number identification service (DNIS) or direct inward dialing (DID). WebSphere Voice Response can receive and use the dialled number if the information is transmitted by the switch as DTMF or MFRI tones in the voice band or as dial pulses on the signaling channel. The dialled number might not be available, depending on the switch to which WebSphere Voice Response is connected and how the trunk interface is configured.

This applies to channel associated signaling (CAS) protocols only.

Use Configuration —> Pack Configuration to configure for dialled number (see Setting address signaling parameters and Setting the telephone numbers for instructions).

Common channel signaling (CCS) Link

Messages can be passed across a common channel signaling link if you are using a protocol such as Signaling System Number 7 or ISDN. The dialled number is provided by a signaling process: either one of those supplied as optional features of WebSphere Voice Response such as SS7 Support for WebSphere Voice Response, or a custom-written one. (For information about writing signaling processes, see the WebSphere Voice Response for AIX: Programming for the Signaling Interface book.

Use Configuration —> Pack Configuration to configure for CCS (see Setting the signaling protocol and Setting the telephone numbers for instructions).

When using a signaling process, the area code is always concatenated to the beginning of the called number. If you do not want this to happen, do not specify an area code.

Exchange data link

If the switch does not support DID or DNIS, and you are using a CAS protocol, the called number can be sent on an exchange data link. The exchange data link can be a direct physical connection between the switch and
the RS-232 serial port on the pSeries computer, or a connection between the switch and WebSphere Voice Response using a local area network and CallPath Server. At the WebSphere Voice Response end, the exchange data link is implemented by a signaling process. You can use a custom-written signaling process, or one of the following exchange data link signaling processes that are provided with WebSphere Voice Response: SMDI, SMSI, VMS, ACL, or CallPath_SigProc.

Fixed or variable length format:

With SMDI, SMSI, and VMS, the called number, calling number (originating number) and the message waiting indication (MWI) number can be transmitted in one of two formats (fixed length or variable length), depending on the switch type and exchange data link signaling protocol selected. With other exchange data links, variable length formats always apply, and no parameters need to be set.

Typically, the switch and the WebSphere Voice Response applications are both configured to handle the same type of format; for example, both expect fixed length. Sometimes however, depending on the switch and your WebSphere Voice Response applications, applications might expect variable length phone numbers, but the switch sends and expects only fixed length numbers. When this happens you can set parameters to ensure that WebSphere Voice Response does the following actions:

• It strips fixed length numbers that it receives from the switch. This action makes the numbers suitable for the applications.
• It pads variable length MWI numbers to a fixed length before it sends them to the switch.

Use **Configuration —> Pack Configuration** (see [Incoming Address Signaling](#) and "Setting the exchange data link parameters” on page 91 for instructions) and then **Configuration —> System Configuration** (see “Setting the exchange data link parameters” on page 91 for instructions).

When using an exchange data link, the area code is always concatenated to the beginning of the called number. If you do not want this to happen, do not specify an area code.

**Channel identification**

Unless you decide to use the System Default Application Profile, you should set up channel identification even if the called number is normally available from the switch. The channel identification can then be used to select an application profile to answer calls for which no called number is unavailable.
Although the channel identification is referred to as “Area Code and Phone Number”, it is not necessarily an area code and phone number. What is important is that it corresponds to the profile ID of an application profile that identifies a state table that can handle the call. For example, in the absence of called number, you might need a state table that asks the caller to choose the required service from a menu.

Therefore, before you can assign values to the Area Code and Phone Number parameters, you need to know what application profile IDs are going to be used. Application profile IDs can be any combination of digits 0 through 9 and the letters A, B, C, and D.

The Area Code is shared by all the channels in a channel group, and is prefixed to the Phone Number. The Phone Number can be unique to each channel, or it can be the same for more than one channel; it depends on which application profile ID you want to associate with that channel.

The composite number (channel group area code concatenated with channel phone number) is handled as the number dialed by a caller only if no other way exists to provide the information.

Use Configuration — Pack Configuration to set up the channel identification (see Setting the telephone numbers for instructions).

Channel identification for exchange data links:

In addition to the area code and phone number, for an exchange data link, you also need to set up a line identifier for each channel.

Planning channel groups

When you know how your applications are going to use the telephony environment, you can plan how to assign the trunk channels to channel groups. This is necessary if you cannot obtain the called number from the switch, but you want calls for different voice response services to be handled by different applications. Assigning channels to channel groups might also be necessary if other differences in signaling characteristics are required.

How many channel groups do you need?

1. Determine how many channel groups will satisfy the requirements of your applications.
   a. Determine which line signaling protocols (for CAS, E&M Wink Start, FXS Loop Start, and so on) are required to establish switch connectivity for each channel group. Think about the switch facilities and other characteristics that are required by the applications you plan to make available (see your “Planning Checklist” in the WebSphere Voice Response for AIX: General Information and Planning book). Compare these
with the “Signaling type templates” on page 575. Make a note of the templates that best match your requirements and choose the most suitable for each application.

**Note:** Some ISDN protocols do not support call transfer. If it not available with your protocol, think about using CAS protocols on some channels, for applications that need to offer call transfer.

b. **CAS only:** Determine whether channels are to allow incoming or outgoing calls, or both. This is known as *direction*. If you want to restrict some channels to either incoming or outgoing, you will need at least two channel groups. Although your applications might receive incoming calls only when in production, you might need to make outgoing calls while the applications are under development.

To specify direction on CCS protocols you must configure the switch lines on the switch side.

c. You can use a channel group to specify the same *area code* for several channels.

d. For each direction, area code, and protocol, you need one channel group.

2. Decide how many channels to dedicate to each application.

3. Allocate the channels on each trunk to channel groups. 30 channels are available on an E1 trunk, and 24 channels are available on a T1 trunk. The total number of channels is limited by your WebSphere Voice Response license. To maximize availability, you might want to divide the channels between trunks for a specific application.

4. Assign channel identification (phone numbers) to each channel.

[Figure 2 on page 40 shows an example of steps 1 on page 38 to 3, with 24 T1 channels on two trunks. Figure 3 on page 41 shows an example of how applications are allocated to channels in one of the channel groups.]
**Step 1**
Determine how many channel groups will satisfy the requirements of your applications.

These applications require 4 different kinds of line signalling, direction, area code and other channel characteristics.

**Step 2**
Decide how many channels to dedicate to each application.

- **Channel Group 1**: 12 channels required to run applications 1 and 2
- **Channel Group 2**: 30 channels required to run applications 3, 4, 5, and 6
- **Channel Group 3**: 3 channels required to run application 7
- **Channel Group 4**: 3 channels required to run applications 8 and 9

**Step 3**
Allocate the channels on each trunk to channel groups.

- **Trunk 1** (24 channels):
  - Channels 1 to 6
  - Channel Group 1
  - Channel Group 2
  - Channels 7 to 24

- **Trunk 2** (24 channels):
  - Channels 1 to 6
  - Channels 7 to 18
  - Channel Group 3
  - Channel Group 4
  - Channels 19 to 21
  - Channels 22 to 24

*Figure 2. Example of assigning channels to channel groups*
**Step4**
Assigning channel identification (phone numbers) to each channel.

- **Trunk 1**
  - 8881234
  - 8881234
  - 8881234
  - 8885555
  - 8885555
  - 8885555

- **Channel Group 1 (Area Code: 123)**
  - 8881234
  - 8881234
  - 8881234
  - 8885555
  - 8885555
  - 8885555

- **Trunk 2**

---

*Area Code and Phone number are used together to find the application profile*

---

**Figure 3. Example of allocating applications to channels**
Chapter 3. Defining the telephony environment using Pack Configuration or wvrteleconf

This chapter includes the following sections:

- "Configuring a pack" on page 44
- "When to configure packs" on page 44
- "Differences between using the wvrteleconf utility and the Pack Configuration Menu" on page 44
- "Parameters set when configuring a pack" on page 45
- "Configuring the telephony environment using the Pack Configuration menu" on page 45
- "Conventions used by wvrteleconf" on page 60
- "Information you need to provide to configure the packs in your system when using wvrteleconf" on page 62
- "Configuring the telephony environment using wvrteleconf" on page 64
- "Example configurations" on page 67

Configuring a pack

Telephony packs in WebSphere Voice Response, can be configured either using:
- The Pack Configuration option on the Configuration menu, or
- The wvrteleconf utility

Both of these methods allow you to configure your telephony environment, whether or not you have knowledge of the various telephony protocols. Each method uses information about the telephony protocols for a large number of countries and switches. For example, the call progress tone values, used for outbound dialing, are set to the specifications used by your country’s network or your PABX.

To configure telephony packs, you will need to enter some details about:
- The country or region
- The switch
- The number of channels
- The signaling protocol to be used

This information is used to select values for system parameters.
The wvrteleconf utility is an ASCII version of Pack Configuration and is designed to be accessible to screen reader users. There are, however, some differences (see “Differences between using the wvrteleconf utility and the Pack Configuration Menu”). The utility also works well with speech input.

Like Pack Configuration, wvrteleconf provides:

- A simple way of configuring a system for the first time (this is referred to as wizard mode).
- Default values wherever suitable.
- A view of the current configuration.
- Direct access to all settings, in order to change them (this is referred to as menu mode).
- Validation before you save your changes, to prevent you saving an invalid configuration.
- A browse version of the menu mode, which allows you to change settings but does not allow you to save any changes.

When to configure packs

Pack Configuration or wvrteleconf can be used to define a working system at most sites, but it is possible that it does not include all possible combinations of telephony hardware and protocols. If configuration of the packs fails to provide a working system at your site, someone who understands telephony must then review and update individual system parameters by using Configuration —> System Configuration (see Chapter 4, “Defining the telephony environment (System Configuration),” on page 73).

Attention

If you use System Configuration to change many of the values that Pack Configuration has set, do not return to Pack Configuration; it might reset some of the settings that you made in Systems Configuration. Use Configuration —> Pack Configuration —> Browse to display the existing values of telephony-related parameters. With Browse, you cannot accidentally change the values.

Differences between using the wvrteleconf utility and the Pack Configuration Menu

Although the wvrteleconf utility is as consistent as possible with the Pack Configuration menu, there are some different features in the wvrteleconf utility:

- There are two separate modes:
  - wizard mode
menu mode

You can save a valid configuration before moving from wizard mode to menu mode.

- To select a country or region type the international dialing code.
- Channel licenses are automatically allocated to trunks. You will, however, need to declare how many licenses you intend to use on the system.
- You can configure more than one trunk at the same time with the same characteristics or copy the configuration of one trunk to configure other trunks.
- In wizard mode, all the channels on a trunk are assigned to one channel group and each explicitly-configured trunk will have a new channel group. You can then use menu mode to split the channels on one trunk into two channel groups.
- Channel group definition and trunk configuration appear separately on the menu. Channels on trunks are allocated explicitly to already-defined channel groups. If you want to allocate channels to an undefined group, you are invited to define the group before proceeding.
- You have the choice of using the same phone number, consecutive phone numbers, or individual phone numbers for all the channels in a channel group.
- You can save a listing of the configuration details to a file.

Parameters set when configuring a pack

Pack Configuration and wvrteleconf set parameters in the Channel, Channel Group, Signaling Type, Exchange Data Link, and Trunk Interface parameter groups. All necessary telephony parameters are set. Both methods of configuring packs select suitable values for several telephony parameters by copying data from templates supplied with WebSphere Voice Response. For details of the actual parameter values in each template, see Appendix B, “System parameter templates,” on page 575.

Configuring the telephony environment using the Pack Configuration menu

Configuring a telephony pack starts at the Pack Configuration window, which displays the basic hardware and software setups for each pack that can be installed in your pSeries computer. The packs are numbered 1, 2, 3, and so on. See the WebSphere Voice Response for AIX: Installation book for more information about the way the packs are labeled.

If you intend to increase the number of channels you use on your system, check whether your WebSphere Voice Response license allows you to use the amount of channels that you are planning. If you are using trunks on a DTTA, check your DTTA trunk license.
**Procedure**

1. From the Welcome window, select **Configuration —> Pack**
   **Configuration —> Change**

2. **Setting the country or region**: The country or region for which WebSphere Voice Response is currently configured is shown to the right of the **Country or Region** button at the top of the window. If the country or region is not correct, click this button.
   The country or region selection window is displayed:

3. Click the required country or region, then **OK** to verify your selection.
   The selected country or region is displayed at the top of the Pack Configuration window. If the name of your country or region is not there, select **Other (E1)** or **Other (T1)**.

4. **Assigning channels and setting the signaling mode**: The Trunk Parameters buttons show the type of signaling protocol for each pack. To change the type of signaling protocol for a pack, click the related button.
The Trunk Interface Parameters window is displayed:

5. Click the drop-down button labeled **Channels Assigned** and select from it the number of channels that you want to assign to this trunk. Remember that your license allows you to use no more than a specified number of channels.

6. Click the drop-down button labeled **Trunk Signaling Mode** and select from it the correct signaling protocol. If you select **ISDN** on an E1 system, the **CAS** button is replaced by **ISDN**. On a T1 system, the NFAS parameters are displayed. For more information about NFAS, see the *WebSphere Voice Response for AIX: General Information and Planning* book.

If you are planning to use NFAS:

a. Select the checkbox labeled **Non Facility Associated Signaling (NFAS) support**
   The NFAS parameters are activated:
b. In the **Trunk ID** field, type in the trunk identifier that your network provider has allocated. The trunk identifier must be different for each NFAS trunk within a trunk group.

c. Click the **Trunk Group** drop-down button and select an NFAS group (1, 2, 3 or 4). The trunk is allocated to the group you select. You can have a maximum of 10 trunks in a group and a maximum of four groups.

d. Click the drop-down button labeled **Signaling Channel**. Select *Primary* if this trunk will be used to carry the signaling information or *Backup* if it will be used to carry signaling information if the primary trunk is out of service. Select *None* if this trunk will be used to carry voice only.

If you are using a common channel signaling protocol other than ISDN, including WebSphere Voice Response’s SS7 support, click **CCS-SP**. (The SP refers to a custom-written signaling process, which you will specify in the **Line Signaling** window.)
7. Click **OK** to confirm your selection. If you have increased the number of configured channels, WebSphere Voice Response displays a Confirm Request window:

![Confirm Request Window](image)

**Note:** If you have increased the number of configured channels on a DTTA, the window also prompts you to check the number of trunks allowed on your DTTA license.

8. When you have checked your license and are sure that it allows you to use the total you have requested, click **Yes, these channels are licensed**.

9. If you cannot check your license, or find that it does not allow you to use the total you have requested, click **Cancel**. Check whether all your licenses have been properly enrolled and distributed. See the *WebSphere Voice Response for AIX: Installation* book for instructions. You might need to contact your IBM Representative and order more channels or, if possible, decrease the number of channels on another trunk, and then assign them to this trunk.

10. **Setting the switch type:** The **Switch Type** buttons show the type of switch to which the pack is connected. If the button shows the wrong type of switch, and the button is enabled, click it.

    The Switch Type window is displayed. (The list of switch types is determined by your choice of **Trunk Signaling Mode** in the Trunk Interface Parameters window.)

**Note:** If you are using FXS with the Nortel DMS100 switch, click **Channel Bank**.
11. Click the required type, then OK to confirm your selection. Your choice of switch type controls the choices you are offered when you select other buttons on the Pack Configuration window. Switch type is not used by the WebSphere Voice Response system. If your switch type is not listed, click Default. The button is reset to the required switch type.

12. **Setting the initial operating status of the pack:** The Operating Status determines the state into which the pack is placed whenever WebSphere Voice Response is started. To change the operating status, click the **Operating Status** button.
The operating status selection window is displayed:

13. Click the required operating status. Possible values are:

   **Inservice**
   The required microcode is loaded and diagnostics are run (this takes about a minute) and the pack is ready to process calls. The
channels are set to Inservice; ready to make or receive calls.
Inservice is the normal setting when you are using the system in production.

**Enabled**
The required microcode is loaded and diagnostics are run.
Although the pack shows as In Service, it cannot be used until you put the relevant channels In Service (**Operations —> System Monitor**).

**Available**
The digital trunk adapter is present but is not ready to communicate with the trunk. The trunk is not ready to process calls until you enable the pack and set the channels In Service (**Operations —> System Monitor**). You can configure the relevant pack by using the buttons to the right of this one. The required microcode, however, is not loaded.

**Defined**
The pack and the associated configuration options are fully disabled. **Defined** is a place holder until the digital trunk adapter is physically installed in the pSeries computer.

14. Click **OK** to confirm your update.
   The **Operating Status** button displays your selection.

15. **Allocating channels to channel groups:** The two **Channel Group** buttons for each pack show which of the 24 T1 or 30 E1 channels on the trunk are allocated to which channel groups.
   If, for example, a channel group button displays **Group 4 (1-10)**, this means that channels 1 to 10 have been allocated to channel group 4.
   Two **Channel Group** buttons are provided so that you can allocate some channels to one channel group and some to another. You might want to do this if some channels are to be used exclusively for incoming calls and some exclusively for outgoing calls. For further information on Channel Groups, see “Supporting one or more protocols” on page 29 and “Planning channel groups” on page 38.

   **Note:** If you want to divide the channels on one trunk to more than two channel groups, you must complete the definition of the two groups you have now started on, save the configuration, close the Pack Configuration window, and then open it again. You can then define the characteristics of two more channel groups on this trunk.

   To change the existing allocation of channels to channel groups, click either of the **Channel Group** buttons. (The buttons are grayed out if the operating status is set to **Defined**.)
The Channel Group Selection window is displayed, showing both channel groups:

![Channel Groups](image)

16. Enter or update channel allocations as required:
   • To allocate all the channels on a trunk to one channel group, enter the lowest and highest channel numbers, and the required channel group number, on a single line. Leave the other line blank by deleting any data shown.
   • To allocate some channels to one group and some to another, ensure that the two ranges of channels do not overlap. For example, allocate channels 1 through 12 to one group and 13 through 24 to the other.

17. Click OK to confirm the update.
   You are returned to the main Pack Configuration window. Your new channel assignments are now shown on the Channel Group buttons.

18. **Setting the signaling protocol:** The signaling protocol that is to be used by the range of channels displayed on the upper Channel Group button is shown on the upper Line Signaling button. The signaling protocol that is to be used by the range of channels displayed on the lower Channel Group button is shown on the lower Line Signaling button.

   **Note:** Changing the signaling protocol affects every channel that is allocated to this channel group, including those on other trunks.
   To change the protocol, click the appropriate Line Signaling button.
   The signaling protocol selection window is displayed. (The list of protocols is determined by your choice of Trunk Signaling Mode and Switch Type and is automatically limited to those supported by your telephony hardware.)
19. Click the protocol that you intend to use. If you have selected CCS-SP as the Trunk Signaling Mode, this window displays CCS protocols that are supported by WebSphere Voice Response, for example SS7_ISUP, and the names User1 through User19. Click the name of the signaling process that you intend to use.

20. Click OK to confirm your selection.

You are returned to the main Pack Configuration window; the Line Signaling button is reset to the required signaling protocol.

21. Setting call direction: The Direction & Tones buttons show the direction settings of the two channel groups. To change the channel group’s call direction, click the appropriate button.

The Call Direction window is displayed:

22. CAS only: The Direction button determines whether WebSphere Voice Response can both make and receive calls. Possible values are:

- **Outgoing**: WebSphere Voice Response can make calls but does not answer incoming calls.
- **Incoming**: WebSphere Voice Response can only receive calls; it cannot make them.
- **Bothway**: No restriction has been put on this channel group.

23. Setting address signaling parameters: For channel associated signaling, the Address Signaling buttons show the type of incoming address
signaling for the two channel groups. To change the channel group's address signaling, click the related button.

**Note:** A change in the address signaling parameters affects *every* channel in this channel group, including those on other trunks.

The Address Signaling window is displayed:

![](image)

24. **Outgoing Address Signaling:** The Outgoing Address Signaling Type button shows the type of in-band signaling that WebSphere Voice Response should use when making an outgoing call.

   *Register* signaling is supported by E&M and DID protocols. Click one of the following:
   - **Register (DTMF):** DTMF signaling
   - **Register (MFR1):** MFRI signaling
   - **Register (Pulse):** Pulse signaling
   - **Register (FGpD):** Feature Group D with MFRI signaling.

   Click the button and drag the mouse pointer to the related value.

25. **Incoming Address Signaling:** The Incoming Address Signaling Type button shows the type of in-band signaling that is used to identify incoming calls. If your switch and protocol do not support incoming address signaling, click None.

   *Register* signaling is supported by E&M and DID T1 protocols and by R2, U.K. Tie, U.K. CallStream, and Italy E1 protocols. Click one of the following:
   - **Register (DTMF):** Fixed length register with DTMF signaling
   - **Register (MFR1):** Fixed length register with MFRI signaling
   - **Register (Pulse):** Fixed length register with pulse signaling
   - **Register (FGpD):** Feature Group D with MFRI signaling.

   For the switch types with which you can use an exchange data link to handle the incoming address signaling, the following options are also offered:
   - **Sig Proc (ACL)
   - **Sig Proc (SMDI)**
• Sig Proc (SMSI)
• Sig Proc (VMS).

If the type of exchange data link signaling process that you want to use is not listed, you must use System Configuration to specify it.

26. If you choose Register (DTMF), Register (MFR1), or Register (Pulse) for incoming address signaling, you must specify the length of the register in the **Register Length** field:

![Address Signalling Table]

27. If you choose one of the **Sig Proc** options, you can then set the parameters for the exchange data link. Choose the **Data rate**, **Parity**, **Communication port** and **Line ID** that are correct for your switch.

**Note:** Parameters that are inappropriate for the **Sig Proc** are not available for selection.
28. **Stripping or padding numbers:** If you selected an **Incoming Address Signaling Type** option of SMSI, SMDI, or VMS, you can choose to strip or pad the phone numbers as required by your switch, WebSphere Voice Response applications, and the selected protocol. For more on using this process, see “Exchange data link” on page 28.

For SMSI/SMDI variable length phone numbers:

- To strip Called or Calling numbers:
  a. Enable the appropriate **Strip** option.
  b. Set the **Minimum** length for the number.
  c. Set the **Character** that is stripped from the numbers received from the switch.

- To pad MWI numbers:
  a. Enable the **MWI Pad** option.
  b. Set the **Length** of the number expected by the switch.
  c. Set the **Character** used to pad each number before it is sent to the switch.

For SMSI/SMDI fixed length phone numbers:
• Disable the **Strip** option. WebSphere Voice Response then uses each number as it is received directly from the switch.

For VMS variable length phone numbers:

• To strip Called or Calling numbers:
  a. Enable the appropriate **Strip** option.
  b. Set the **Minimum** length for the number.
  c. Set the **Length** of the number expected by the switch.
  d. Specify the **Character** that is stripped from the numbers received from the switch.

• To pad MWI numbers:
  a. Enable the **MWI Pad** option.
  b. Set the **Minimum** length for the number.
  c. Set the **Character** used to pad each number before it is sent to the switch.

For VMS fixed length phone numbers:

• Disable the **Strip** option. Enter the number of digits that make up the Called and Calling numbers in **Length**. WebSphere Voice Response returns an error if the numbers it receives is of any length other than the one specified by **Length**.

29. Click **OK** to complete the update.

You are returned to the main Pack Configuration window.

30. **Setting the telephone numbers**: The **Telephone Numbers** button displays the area code and telephone numbers that are associated with the two channel groups. If different channels are set to different numbers, the digits that are not common to all channels are shown as Xs. The **Area code** is the same for all channels in a group.

Whenever the **called number** is not available, the area code and telephone number values are used as **channel identification**, to find an application profile to answer a call. You specify a phone number (and, optionally, an area code) for each channel on your system: the phone number does not have to be unique for each channel. Then you must provide an application profile to match each number.

**Note**: If you are using a signaling process or exchange data link, the area code is used even when the called number is available: it is concatenated to the beginning of the called number. If you do not want this to happen, ensure that the **Area Code** field is blank.

For more information, see [“Answering each call with an appropriate application” on page 30](#) and [“Channel identification” on page 37](#).
This window also shows the line identifiers (Line IDs) that the exchange data link uses. For CallPath_SigProc, set each line ID to the station or phone number for the line; each of the line IDs must be unique.

To change the area code, telephone numbers, or the line IDs, click the appropriate Channel IDs button.

The Channel Identification window is displayed:

For information about how the area code and telephone numbers are used, see “Channel identification” on page 37. For information about message line identifiers, see “Message Info Line Identifier” on page 388.

31. When you have set all the numbers and the area code correctly, click OK to complete the update.

In the Pack Configuration window, the telephone number is displayed if it is the same for all channels. If different channels are set to different numbers, the digits that are not common to all channels are shown as Xs.

32. Configuring other packs: If you have more than one pack installed in your system, repeat Steps 10 on page 49 to 31 for each of the other packs.

33. Saving your changes: An asterisk (*) is displayed next to the pack identifier if you have changed the values of any of the following:
   - Trunk Parameters
   - Switch Type/Resource Type
   - Channel Group/Resources
   - Line Signaling
Attention: If you select the Save checkbox, other system parameter values might be reset to default values, undoing any work done using System Configuration.

If you do not want this to happen, do not select the Save checkbox, even if you haven't made any changes.

34. Click File —> Save.
   In this example, all changes made to pack 1A are saved and many other system parameters are reset. For pack 1B, changes made to Operating Status, Direction, Address Signaling, and Telephone Numbers are saved but no other system parameters are reset.

35. Click File —> Close.
   You are returned to the Welcome window.

The new configuration for each pack takes effect when WebSphere Voice Response is next restarted.

What next?
In most cases, your telephony configuration is now complete. Depending on your requirements, however, you might need to do further configuration. Use the instructions in Chapter 5, “Exchange data links and common channel signaling,” on page 91 and Chapter 6, “Advanced system parameter settings,” on page 101.
When you have restarted WebSphere Voice Response and you are sure that the configuration has worked, make a backup copy of the system configuration (see “Making a backup copy of system parameter values” on page 19).

Your telephony packs are now ready to handle incoming and outgoing calls, but you must also set up application profiles. Refer to the WebSphere Voice Response for AIX: Designing and Managing State Table Applications book for details.

You must also put the telephony packs and channels into service. Refer to the WebSphere Voice Response for AIX: Managing and Monitoring the System book for details.

Conventions used by wvrteleconf

The conventions used by wvrteleconf are:

- “Dialogs”
- “Input” on page 61
- “Retaining the current value” on page 62
- “Navigating” on page 62
- “Help” on page 62

Dialogs

The utility presents one dialog at a time, using ASCII characters on a console screen. The screen is cleared before each dialog. Each dialog begins with a heading, which provides the following information:

- The name of the utility, “Telephony Configuration”.
- The dialog number.
- In some dialogs, the name of the object being configured (for example, trunk one).
- The title of the dialog.

The body of the dialog follows the heading and is followed by a command prompt.

These are the types of dialog:

- “Action menu” on page 61
- “Decision point” on page 61
- “List of values” on page 61
- “Entry field” on page 61
- “Confirmation” on page 61
Action menu

An action menu allows you to choose what to do from a number of options, by entering the number assigned to the option. There is no default action.

Decision point

A decision point is similar to an action menu but there may be a default, which you can select by pressing Enter. There are normally two or three choices in a decision point.

List of values

A list of values allows you to choose a value for an item by entering the number assigned to the value. The list is in the order of the numbers assigned to the values, with one important exception: the current (or default) value can be retained by pressing Enter. The current (or default) value is always listed at the top of the list, rather than in the position it would otherwise occupy. However, the numbers assigned to the values are fixed. For example, if the current value is '7', typing 7 is equivalent to pressing Enter.

Entry field

An entry field is where you enter a single value for an item: this might be one or more characters depending on the requirements for the item. The current (or default) value can be retained by pressing Enter.

Confirmation

You are asked to confirm only very significant or destructive choices. You can press Enter to confirm your choice, or press F3 or Escape followed by 3 to return to the dialog where you made the choice.

Information-only

Information-only dialogs just provide information. Press Enter to continue to the next dialog.

Input

Each dialog expects one input from you. This input may be:

- The Enter key.
- One or two numeric keys from a known list, followed by the Enter key.
- Character data followed by the Enter key.
• A function key.
• Escape followed by numeric key or keys.

The meanings of the function keys and Escape followed by numeric keys are equivalent.

If your input is not understood by the utility as being valid, the dialog is presented again with a message to explain why. If you need further explanation, press F1 or Escape followed by ’1’ to see the help information.

**Retaining the current value**

After you have set a value, this value becomes the current value; to retain this value, press Enter. For some items, there is a default value, and this is treated exactly like the current value. There is no simple way of restoring the default value for a single item.

**Navigating**

Forward navigation is done by pressing Enter. Other navigation is controlled by pressing function keys. On terminals that do not support function keys, press the Escape key followed by the equivalent numeric keys. To list the function keys, press F2 or Escape followed by 2.

**Help**

Help is accessed by pressing F1 or Escape followed by 1. To return from the help to the dialog, press Enter.

---

**Information you need to provide to configure the packs in your system when using wvrteleconf**

This section provides an overview of the information that you need to give the wvrteleconf utility to configure the packs in your system. The information is given in the order in which the wizard collects it. You can alternatively go into the direct-access or menu mode and change the information directly. The types of information you will need are:

- “Country or region”
- “Channel license declaration” on page 63
- “Adapter configuration” on page 63
- “Trunk configuration” on page 63
- “Channel identification” on page 64
- “SIP settings or exchange data link configuration” on page 64

**Country or region**

You must specify the country or region in which the system is to be used by typing its international dialing code. For example, for the U.S. or Canada, type 1; for France, type 33.
There is no default value for country or region. Once you have made your choice, the telephony configuration wizard excludes choices of values for other items that are inappropriate for the country or region that you have selected.

Parameters that seriously affect the operation of the system are set according to the country or region selected; this might affect compliance with telecommunications authority regulations, and must only be done by authorized personnel familiar with these requirements.

The choice of country or region dictates whether the system is E1 or T1. In an E1 system, each trunk has 30 channels; in a T1 system, each trunk has 24 channels.

Channel license declaration
You must declare the number of channels you intend to use, and are licensed to use. There is no default value. It is up to you to check your license agreement and enter the number of channel licenses that you intend to use on this system. These licenses are then allocated to the trunks that are physically attached to your system until all the licenses are used up. For example, if you have two E1 trunks, but declare only 50 licenses, you will be able to use 30 channels on the first trunk, but only the first 20 channels on the second trunk.

Adapter configuration
Each voice response system can have up to four digital trunk adapters physically installed. These can be either DTTAs, or DTEAs. There are no settings that apply to the DTTAs, so you will only see this dialog if you have one or more DTEAs installed.

Trunk configuration
Each voice response system can have up to four digital trunk adapters physically installed. At a minimum, you must configure at least one trunk. The wizard offers default values for all parameters, so if you are unsure what to do, press Enter to retain the current setting.

Parameters that apply to the whole trunk include line signaling protocol, call direction, outgoing address signaling, incoming address signaling, and area code. The wizard sets these identically for all channels on the trunk. If you need to have channels with different attributes on the same trunk, you can later split the channels on the trunk into two groups. See “Defining a channel group using wvrteleconf” on page 67 for more information.

In the trunk specification dialog, you can specify multiple trunks to be configured identically.
**Channel identification**

Channel identification includes phone numbers and, if you have an exchange data link, line identifiers.

Phone numbers allocated to the channels can be used, in the absence of the called number being sent by the switch (using DNIS or CLID), to identify the application profile to invoke an application to answer the call. There is no need to allocate phone numbers to channels if the called number is passed by the switch. The wizard makes it easy to specify one phone number for all the channels on the trunk, or to allocate consecutive numbers.

**SIP settings or exchange data link configuration**

You will only have to configure SIP settings or an exchange data link, if choices you made in one or more trunk configurations indicated that you would be using SIP or an exchange data link. These settings apply to the whole system.

---

**Configuring the telephony environment using wvrteleconf**

This section tells you how to configure the telephony environment on a newly-installed system, or whenever you want to reconfigure one or more trunks.

**Prerequisites for all users**

If you need to change configuration information, you need to know the password for an administrator profile that has change access to Pack Configuration. If you only need browse permission, you need to know the password for an administrator profile that has browse access to Pack Configuration.

You need to know the international dialing code for your country or region. For example, for the U.S. or Canada, the code is 1; for the United Kingdom, the code is 44.

You need to know how many channel licenses you have purchased and intend to use on this system: you need one license for each channel on a trunk (30 on an E1 trunk, 24 on a T1 trunk).

**Prerequisites for screen-reader users**

Many telephony-related acronyms are used: it is preferable to set the screen reader to read out all acronyms.

In some configuration, phone numbers will be read out: it is preferable to set the screen reader to read out individual digits.
Other information

At any time you can:

- Get help on the current dialog by pressing F1 or Escape followed by '1'.
- List the function keys by pressing F2 or Escape followed by '2'.
- Return to the previous screen by pressing F3 or Escape followed by '3'.
- View a summary report on the screen by pressing F9 or Escape followed by '9'.
- Quit by pressing F10 or Escape followed by '0'. (If you have made any changes you will be asked if you want to validate and save them).
- It is preferable that you have experience of configuring telephony systems to use the menu mode. Selecting Configure all settings when in the menu mode will guide you through the settings in the correct order.

Procedure for configuring a telephony pack

1. Log on as dtuser and type the following command:
   wvrteleconf
2. The first two dialogs ask you for the administrator name and password. Type this information and press Enter.
3. When your system is completely new and unconfigured, this takes you straight into wizard mode. If you have already configured one or more trunks, however, the mode choice dialog offers you the choice of wizard or direct-access modes.
4. Press Enter to continue from the 'wizard setup' dialog.
5. When you system is completely new and unconfigured, you must provide information about the country or region, and the number of channels you intend to use. See "Country or region" on page 62 and "Channel license declaration" on page 63 for information.
6. If you have DTEA adapters installed, you need to provide some information about them. See "Adapter configuration" on page 63 for more details.
7. The wizard then presents you with the 'trunk selection' dialog. Here you must select the trunk or trunks you want to configure. You can configure as many trunks identically as you want. Enter the trunk numbers separated by commas. See "Trunk configuration" on page 63 for more information.
8. Depending on the protocols you have configured, you may have to specify channel identification for the channels on the trunk. See "Channel identification" on page 64 for more information.
9. If further trunks are available to be configured, the wizard asks if you want to configure another trunk.
10. If you answered “yes” to the previous question the wizard presents the **trunk selection** dialog again. You can leave the configuration of the remaining trunks until later. You can only use the trunks you have already configured.

11. If you select a trunk, the next dialog asks you if you want to copy a previously-configured trunk. If you elect not to copy, you will go through the trunk configuration dialogs again.

12. Depending on the protocols you have configure, you may have to specify system-wide SIP or exchange data link settings. See “SIP settings or exchange data link configuration” on page 64 for more information.

13. When you reach the finish dialog, validate and save all your changes.

14. If your configuration is valid, you can now exit safely, optionally creating a summary report.

15. If your configuration is not valid, the validation result dialog tells you what it found wrong. Validation proceeds in the same direction as the wizard, and the validation results are presented one by one in that order. You can fix each validation result by pressing Enter in the validation result dialog. You have two alternatives:
   a. If you realize that the problem is with the current trunk configuration, and not the trunks you configured earlier, you can discard the changes on the current trunk.
   b. You can go into menu mode to make your own changes. Although this might create further inconsistencies. See Making changes to the configuration using wvrteleconf.

**Procedure for browsing a configuration using wvrteleconf**

This procedure instructs you how to browse the configuration details, without being allowed to save any changes you make.

1. Log on as dtuser and type the following command:

   ```
   wvrteleconf
   ```

2. The first two dialogs ask you for the administrator name and password. Type this information and press Enter.

3. You can now select **Browse configuration details**.

**Procedure for making changes to the configuration using wvrteleconf**

This procedure instructs you how to make changes to the configuration using the menu mode of the wvrteleconf utility.

1. Log on as dtuser and type the following command:

   ```
   wvrteleconf
   ```

2. The first two dialogs ask you for the administrator name and password. Type this information and press Enter.

3. You can now select **Change configuration details**.
Defining a channel group using `wvrteleconf`

This section provides information about channel groups, to which you can allocate some of the channels on a trunk.

A channel group is a set of channels that share the same characteristics. By default all the channels on a trunk belong to the same group. But to have channels with different characteristics on the same trunk, you can allocate some to one group and some to another. There are 16 channel groups, numbered 1 through 16.

Each channel group can use a different line signaling protocol, call direction, incoming or outgoing address signaling protocols, and area code. For example, if the Trunk Interface type is T1, one channel group can use the E&M trunk protocol and another the FXS trunk protocol. You could have two channel groups using FXS, one to support incoming calls, and another to support outgoing calls.

If you require channels with different mixtures of line signaling protocol, call direction, incoming and outgoing address signaling protocols, and area code, you need to plan your channel groups carefully before allocating all the channels.

The channels in a channel group can belong to multiple trunks, so changing the channel group setting changes the characteristics of all the channels in a the group, whichever trunk they belong to.

New channel groups can only be defined in the `trunk configuration` menu, as channel groups must be assigned to a trunk before they can be edited. Existing channel groups can be updated through the `channel group` menu, or in the `trunk configuration` menu.

Example configurations

If you are configuring multiple packs, and particularly if you are planning to use more than one protocol, use the following examples as a guide:

- “Example 1: T1 mixed system” on page 68 shows a system using CAS, ISDN (without NFAS), and another CCS protocol.
- “Example 2: T1 ISDN non-facility associated signaling (NFAS)” on page 69 shows a system using CAS and ISDN BCS34.
- “Example 3: E1 mixed system” on page 70 shows a system using CAS and Euro-ISDN.
**Example 1: T1 mixed system**

You can mix CAS, ISDN, and another common channel signaling protocol on the same WebSphere Voice Response system. However, these cannot coexist on the same trunk at the same time so you must create a channel group for each protocol, as shown in Figure 4.

A maximum of 16 trunks (384 channels) can be configured on a T1 system.

**Channel associated signaling (CAS)**

The default Line Code (AMI/ZCS) and Framing Mode (D3/D4) are assumed. To use B8ZS Line Code or ESF Framing Mode, you must use System Configuration parameters to set them (see “T1 line code, framing mode, and framing format” on page 120).

![Figure 4. T1 mixed CAS and CCS protocols](image)

**ISDN without NFAS**

The switch to which WebSphere Voice Response is connected must be configured as network side, because WebSphere Voice Response can operate only as user side.

Only one version of ISDN signaling can be used in the system at any time. For example, with an AT&T 5ESS switch, you cannot run 5E8 user side on one trunk and 5E9 user side on another trunk.
Other common channel signaling (CCS)

Select the Default Switch Type, because the custom-written signaling process software might support any range of switches.

The channels on a trunk can be allocated to more than one channel group. For example, on trunk 5, channels 1 through 12 could be allocated to group 5 and use signaling process User5, while channels 13 through 24 could be allocated to group 6 and use signaling process User6.

The default Line Code (AMI/ZCS) and Framing Mode (D3/D4) are assumed. If you need to use B8ZS Line Code or ESF Framing Mode, you must use System Configuration parameters to set them (see “T1 line code, framing mode, and framing format” on page 120).

Example 2: T1 ISDN non-facility associated signaling (NFAS)

An ISDN NFAS system can be configured with channel associated signaling (CAS) or common channel signaling (CCS) trunks also present, although this configuration is not usual.

You can define up to four NFAS groups with a maximum of 10 trunks in each. For each NFAS group you need to specify one primary trunk to carry the signaling for all the trunks in the NFAS group. Channel 24 on the primary trunk becomes a D-channel carrying signaling information and all other channels in the group become B-channels carrying voice.

For some switches and line signaling protocols you can also specify a backup trunk, see WebSphere Voice Response for AIX: General Information and Planning for a list of switches and protocols that have the D-channel backup facility
enabled. If the primary D-channel goes out of service or becomes alarmed, a backup D-channel in standby mode will automatically take over the transport of signaling information. The backup D-channel does not carry voice.

**All trunks**

You must specify the same **Switch Type** and **Line Signaling** values for each trunk.

Each trunk must have a unique trunk identifier, which is specified in your switch configuration and must be matched correctly in the **Trunk Interface Parameters** window for each trunk (see Non Facility Associated Signaling (NFAS) support). The trunk identifiers are the numbers that are shown in parentheses on the **Trunk Parameters** buttons.

Removing or adding adapters can change the way trunks are numbered. Make sure your configuration identifies the correct physical trunks.

**The primary and backup trunks**

In [Figure 5 on page 69](#), pack 3 is identified as the primary trunk by a **P** on its **Trunk Parameters** button. The backup trunk is identified by a **B**.

For the primary and backup signaling trunks, B8ZS Line Code and ESF Framing Mode are automatically set because these are the only valid options (see “T1 line code, framing mode, and framing format” on page 120).

**The non-signaling trunks**

T1 trunks on ISDN always default to B8ZS Line Code and ESF Framing Mode.

**Example 3: E1 mixed system**

You can mix CAS, ISDN, and other common channel signaling protocols on the same WebSphere Voice Response system. However, these cannot coexist on the same trunk at the same time so you must create a channel group for each protocol, as shown in [Figure 6 on page 71](#).

A maximum of 16 trunks (480 channels) can be configured on an E1 system.
Channel associated signaling (CAS)

ISDN:

The switch to which WebSphere Voice Response is connected must be configured as network side, because WebSphere Voice Response can operate only as user side.

Click the Default Switch Type, because Euro-ISDN is supported on many switches.

Other common channel signaling (CCS):

Click the Default Switch Type, because the custom-written signaling process software might support any range of switches.

The channels on a trunk can be allocated to more than one channel group. For example, on one trunk, channels 1 through 12 could be allocated to group 5 and use signaling process User5, while channels 13 through 30 could be allocated to group 6 and use signaling process User6.
Chapter 4. Defining the telephony environment (System Configuration)

This chapter includes the following sections:
- "When to use System Configuration"
- "Parameters used to define channel characteristics" on page 74
- "When to use System Configuration"
- "Defining trunk interfaces" on page 75
- "Defining signaling types" on page 78
- "Defining channel groups" on page 80
- "Defining channels" on page 84
- "Using wvrsysconf to define telephony parameters" on page 86

Figure 7 on page 75 shows how the parameter groups are used to define the characteristics of trunks and channels. These parameters are collectively known as the telephony parameters.

When to use System Configuration

You can use these procedures, either to set up your system the first time, or to reset any of the parameters to specific values, at any time. If you need an explanation of any of the parameters, see “System parameters reference” on page 195, or select Help when you are in the parameter window. For general instructions on setting parameter values, see “Setting the value of a system parameter” on page 13.

However, it is better that you use Pack Configuration first and use System Configuration only to fine-tune individual system parameters. This particularly applies to ISDN, and other common channel signaling protocols. (See Chapter 3, “Defining the telephony environment using Pack Configuration or wvrteleconf,” on page 43.)

Note: There are two interfaces provided for performing system configuration:
- The System Configuration option on the Configuration menu
- The wvrsysconf command-line utility

Both of these methods allow configuration of all the telephony parameter types described in “Parameters used to define channel characteristics” on page 74, but wvrsysconf is additionally designed to be accessible to users of
Parameters used to define channel characteristics

Because channels physically belong to trunks and logically belong to channel groups, the characteristics of each channel are defined with several different groups of parameters:

- Channel parameters: specifying unique characteristics of the channel
- Trunk Interface parameters: for the trunk to which the channel is assigned
- Channel Group parameters: for the channel group to which the channel belongs
- Signaling Type parameters: for the signaling type specified for the channel group to which the channel belongs (channel associated signaling only)
- ISDN Signaling parameters: for all channels using ISDN

The way the parameter groups are used to define channel and trunk characteristics is shown in Figure 7 on page 75. If you use System Configuration to define the telephony environment, you need to understand these relationships; if you use Pack Configuration, all you need to know is that the same parameters are being set, but Pack Configuration is keeping track of the relationships for you.

When do the parameter values take effect?

Most of the telephony parameters take effect as soon as you disable then enable the packs that process the channels. Remember that if a channel group definition applies to channels on more than one pack, and you change the definition, all the packs that use that definition must be disabled then enabled.

Attention

For more information, see “When do new values take effect?” on page 16.
Defining trunk interfaces

Use this procedure to define the interfaces between WebSphere Voice Response and the trunks to which it is connected. If more than one trunk connection exists, ensure you define them all. For an introduction, see “Trunks” on page 22.

Procedure

1. From the Welcome window, select Configuration → System Configuration → Change.
2. Click General.
3. **Defining the trunk protocol:** Click **Trunk Interface**.

4. Click the appropriate type of interface. As a rough guide, T1 applies to Canada, China (Hong Kong S.A.R.), Japan, and the U.S.A; E1 to Europe and Latin America. If in doubt, consult your network provider.

5. Click **OK**.

6. **Close** the General parameter group window.

   The system displays the System Configuration window:

7. **Copying a template:** Click **Trunk Interface**.

   WebSphere Voice Response includes several *named* trunk interface templates. Select the named template that most closely suits your requirements and copy it, following the procedure in "Using system parameter templates" on page 17. (For details of the templates, use the tables in "Trunk interface templates" on page 585.) Then return to this procedure to edit the values copied from the template.
Attention: If you do not select the correct template, one of the following conditions might occur:

- You cannot enable the trunk.
- You have enabled the trunk successfully but you cannot communicate with the switch.

8. Displaying the Trunk Interface parameters: Open the numbered trunk for which you want to define the interface.

The system displays the existing values of all the parameters that apply to all channels on the selected trunk.
9. **Changing the Trunk Interface parameters:** Review the explanations of the parameters in “System parameters reference” on page 195 or in the online help. If the default value for any parameter is not correct, change it.

   **Note:** The default value of Operating Status for the first trunk interface is Available, whereas for the other trunk interfaces, the default value is Defined. The normal setting when you are using the system in production is InService.

10. **ISDN:** If you are using Non-Facility Associated Signaling (NFAS) ISDN, you must set the ISDN Trunk Identifier parameter to the trunk identifier that your service provider allocated. The trunk identifier must be different for each trunk.

11. **Close** the Trunk Interface window and click File —> **Save**.

Although the System Configuration window is still displayed, the system has configured the trunk interface.

---

### Defining signaling types

Use this procedure to define the signaling types needed in your telephony environment. **This procedure applies to Channel associated signaling (CAS) only.**

**Note:** You must use the same common channel signaling (CCS) protocol for all the channels on a trunk. Therefore, you do not need a Signaling Type definition when using CCS protocols. If you are using a CCS protocol, skip this section and go to “Defining channel groups” on page 80.

For an introduction to signaling types, see “Supporting one or more protocols” on page 29.

**Procedure**

1. From the Welcome window, select **Configuration** —> **System Configuration** —> **Change**.
2. Click **Signaling Type**.

   **Displaying the signaling type parameters:** WebSphere Voice Response includes numbered and named signaling types. These named signaling types are predefined with default values for various protocols, such as FXS Loop Start. Use the named signaling types as templates and copy them over to the numbered signaling types (1 through 16). The named signaling types must be copied over to the numbered signaling types because they cannot be edited. To copy them over, use the procedure in “Using system parameter templates” on page 17. Then follow this procedure to edit the
values copied from the template:

3. **Open** the numbered signaling type you need to edit.
   The system displays the existing values of the parameters:

![Signaling Type Table]

4. **Dialled number information:** If you plan to use dialled number information that is transmitted across the trunk during call setup (see "Configuring WebSphere Voice Response to get the called and calling numbers" on page 35), set the parameters as shown in **Table 6 on page 80**.
5. **Hangup Detection**: If you plan to use far-end hangup detection (in the absence of signaling information from the switch, for example, if you are using FXS Loop Start or RE Signaling), you must set either Constant Energy Detection or Cadenced Detection for the Hang Up Detection parameter. You might need to set the values of some of the other parameters in this group (see “Setting parameters for hangup tone detection” on page 112).

6. **Anything else?**: Review the explanations of the remaining parameters in “System parameters reference” on page 195 or in the online help. If the default value for any parameter is not correct, change it.

7. **Close** the Signaling Type window and click **File —> Save**.

   Although the System Configuration window is still displayed, the system has saved the signaling type.

**Table 6. Dialed number information parameters**

<table>
<thead>
<tr>
<th></th>
<th>Values for fixed length incoming register signaling</th>
<th>Values for feature group D incoming register signaling</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Incoming address</strong></td>
<td>Set this parameter to <strong>Fixed Length</strong>.</td>
<td>Set this parameter to <strong>Feature Group D</strong>.</td>
</tr>
<tr>
<td><strong>register type</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Register length</strong></td>
<td>Set this parameter to the number of digits actually provided by the switch. In general, a PABX sends the extension number only; a central office is likely to send more digits. The default value is 5.</td>
<td>Not used.</td>
</tr>
</tbody>
</table>

**Defining channel groups**

Use this procedure to define the channel groups that are needed in your telephony environment. For an introduction to channel groups, see “Supporting one or more protocols” on page 29.

**Procedure**

1. At the Welcome window, select **Configuration —> System Configuration —> Change**.
2. Click **Channel Group**.
3. **Displaying the Channel Group parameters**: Open the first channel group to be defined.
   The system displays the existing values of the parameters that apply to all channels assigned to this group.
4. If you have specified a channel associated signaling protocol, go to Step 7 on page 82.

5. **Common Channel Signaling protocols:** If you have specified a common channel signaling protocol (ISDN or a custom-written signaling process), set the parameters as shown in Table 7.

<table>
<thead>
<tr>
<th>Table 7. Common channel signaling parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Parameter</strong></td>
</tr>
<tr>
<td>Call information type</td>
</tr>
</tbody>
</table>
   | Signaling process type | Set this parameter as appropriate:  
   |  | • The ISDN... value for your ISDN protocol  
   |  | • User1 through User19: the name of your custom-written signaling process. |
   | Area code | Set this to a 1- to 6-character value. |
   |  | In the event that called number information is not received, the Area Code value is used with the Phone Number parameter of the channel, to create the channel identification to be used to find the application profile. |
6. Go to [Anything else?](#)

7. **Channel Associated Signaling Protocols:** Click the numbered **Signaling Type** that you defined when following the procedure "Defining signaling types" on page 78. Do not click any of the named signaling types because you cannot edit them later.

8. Click the start type, using the parameter appropriate to your protocol:
   - T1 CAS Protocols:
     - DID Start Type
     - FXS Start Type
     - E&M Start Type
   - E1 CAS Protocols:
     - UK Tie/DDI Start Type
     - FXS Start Type (Loop Start only)

9. Depending on what you decided in "Configuring WebSphere Voice Response to get the called and calling numbers" on page 35, follow the steps in one of the following tables:
   - Table 8
   - Table 9 on page 83
   - Table 10 on page 83

**Table 8. Dialed number information parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Values for fixed length incoming register signaling</th>
<th>Values for feature group D incoming register signaling</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call information type</td>
<td>Set this parameter to <strong>Register.</strong></td>
<td>Set this parameter to <strong>Register.</strong></td>
</tr>
<tr>
<td>Incoming address signaling type</td>
<td>Set this parameter to MFR1, DTMF, or Dial Pulse, depending on the capabilities of your switch. Note that the incoming and outgoing address signaling types are normally the same.</td>
<td>Set this parameter to <strong>MFR1.</strong></td>
</tr>
<tr>
<td>Signaling process type</td>
<td>Set this parameter to <strong>None.</strong></td>
<td>Set this parameter to <strong>None.</strong></td>
</tr>
</tbody>
</table>
Table 8. Dialed number information parameters (continued)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Values for fixed length incoming register signaling</th>
<th>Values for feature group D incoming register signaling</th>
</tr>
</thead>
<tbody>
<tr>
<td>Area code</td>
<td>Set this to a 1- to 6-character value.</td>
<td>Set this to a 1- to 6-character value.</td>
</tr>
</tbody>
</table>

In the event that called number information is not received, the Area Code value is used with the Phone Number parameter of the channel, to create the channel identification to be used to find the application profile.

Table 9. Channel identification parameters

<table>
<thead>
<tr>
<th>Call information type</th>
<th>Set this parameter to None, which is the default.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Signaling process type</td>
<td>Set this parameter to None, which is the default.</td>
</tr>
<tr>
<td>Area code</td>
<td>Set this to a 1- to 6-character value.</td>
</tr>
</tbody>
</table>

In the event that called number information is not received, the Area Code value is used with the Phone Number parameter of the channel, to create the channel identification that is to be used to find the application profile.

Table 10. Exchange data link parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Values SMSI, SMDI or VMS</th>
<th>Values for ACL</th>
<th>Values for CallPath_SigProc</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call information type</td>
<td>Set this parameter to Signaling Process.</td>
<td>Set this parameter to Signaling Process.</td>
<td>Set this parameter to Signaling Process.</td>
</tr>
<tr>
<td>Signaling process type</td>
<td>Set this parameter to SMSI/SMDI/VMS.</td>
<td>Set this parameter to ACL.</td>
<td>Set this parameter to User1.</td>
</tr>
</tbody>
</table>
### Table 10. Exchange data link parameters (continued)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Values SMSI, SMDI or VMS</th>
<th>Values for ACL</th>
<th>Values for CallPath_SigProc</th>
</tr>
</thead>
<tbody>
<tr>
<td>Area code</td>
<td>Set this parameter to the 1- to 6-character prefix that is to be used for all channels in the channel group. The area code is concatenated to the dialed number sent on the exchange data link.</td>
<td>Set this parameter to the 1- to 6-character prefix that is to be used for all channels in the channel group. The area code is concatenated to the dialed number sent on the exchange data link.</td>
<td>Set this parameter to the 1- to 6-character prefix that is to be used for all channels in the channel group. The area code is concatenated to the dialed number sent on the exchange data link.</td>
</tr>
</tbody>
</table>

**Anything else?** Review the explanations of the remaining parameters that are listed in "Channel parameter group" on page 183 or in the online help. If any of the values are not suitable for use with your switch, change them.

Although the System Configuration window is still displayed, the system has saved the parameter values for the channel group.

**Creating further groups**

When you have defined a channel group, you can use that definition to create other channel groups. To copy a channel group definition, follow the procedure in "Using system parameter templates" on page 17. Then return to this procedure to edit the definitions as appropriate.

Close the Channel Group window and click File —> Save.

**Defining channels**

Use this procedure to define the individual telephony channels. For an introduction to channels, see "Channels" on page 23.

**Procedure**

1. **Parameter:** At the Welcome window, select Configuration —> System Configuration —> Change.
2. Click Channel.
3. **Displaying the Channel parameters:** Open the channel. The channel buttons are labeled with the trunk identifier and the channel number (for example on the button labeled 01/02, 01 means trunk 1 and 02 means channel 2).

The system displays the existing values of all the parameters that apply to that channel.

4. **Assigning the channel to a channel group:** Every channel must be assigned to a group. Set the Channel Group to the correct channel group number.

5. **Dialled number information:** If you plan to use dialled number information (see "Configuring WebSphere Voice Response to get the called and calling numbers" on page 35), set the Phone Number parameter to the 1- to 12-character string (digits 0 through 9, letters A, B, C, or D) that is to be used to identify the channel. You can use this string with the area code to retrieve an application profile that calls an application that answers incoming calls on this channel if the dialled number information is not available.

6. **Exchange data link:** If you plan to use an exchange data link (see "Configuring WebSphere Voice Response to get the called and calling numbers" on page 35), you must specify an EDL Message Info Line Identifier (the code that the switch uses to identify this channel when it transmits information via the exchange data link). This identifier must be the same length as the EDL Line Identifier Number Length parameter in the Exchange Data Link parameter group.

   Also, set the Phone Number parameter to the 1- to 12-character string (digits 0 through 9, letters A, B, C, or D) that is to be used to identify the channel. You can use this string with the area code to retrieve an application profile that calls an application that answers incoming calls on this channel if the dialled number information is not available.

7. **Channel identification:** If you plan to use channel identification to determine which application is to answer a call (see "Channel identification" on page 37), you must set the Phone Number parameter to the 1- to 12-character string (digits 0 through 9, letters A, B, C, or D) that is to be used to identify the channel. You can use this string with the area code to retrieve an application profile that calls an application that answers incoming calls on this channel.
8. **Copying the channel definition to define further channels**: When you have defined a channel, you can use that definition to create other channel definitions. To copy a channel definition, follow the procedure in "Using system parameter templates" on page 17. Then return to this procedure to edit the definitions as appropriate.

9. Close the Channel window and click **File —> Save**.

Although the System Configuration window is still displayed, the channel has been defined.

**Using wvrsysconf to define telephony parameters**

The wvrsysconf utility provides direct manipulation of the telephony parameters that are used in system configuration. It does this by exporting the parameters and their values in the form of an XML document. It also creates an XML Schema Definition (XSD) file that contains any restrictions on the permitted values for the parameters.

Figure 8 shows how wvrsysconf creates Document Object Model (DOM) representations of the telephony parameters held in the WebSphere Voice Response database, and then exports these to the XML data file (wvrSysConf.xml is the default name), and the XML schema definition file (wvrSysConf.xsd).

**Note**: To be able to run wvrsysconf, you must first download and install the xml4c5_4_0-aix_510-xlc_50.tar.gz libraries from the alphaworks website at http://www.alphaworks.ibm.com/aw.nsf/download/xml4c.
You can edit the XML data file (using your preferred editor) to change the required parameters to the values you want. Once complete, you can then validate the file against the permitted values that are held in wvrSysConf.xsd, using a standard XML/XSD validation tool, such as that provided in WebSphere Developer Studio.

Once you have a valid XML document, you can use wvrsysconf again to import the document into the WebSphere Voice Response database. At the time of the import, a new schema definition file is created, and this is used to re-validate the XML data. This process is shown in Figure 9.

Note:
1. wvrSysConf.xsd is a read-only file and cannot be imported; the definitions it contains cannot be changed by using wvrsysconf.
2. Some WebSphere Voice Response parameters use rules that represent constructs such as “Maximum volume must be greater than minimum volume”. These rules cannot be expressed in XML schema syntax, and so their validation is done during the import, after the main validation of the parameter data against the schema definitions. Values that are only valid in stepped increments (for example 5, 10, 15, 20) also cannot be expressed in schema syntax. Such values are rounded down to the nearest increment on import, after the validation of the parameter data against the schema definitions. If rounding does occur, a warning is displayed.

Definition of wvrsysconf.xsd
The XML schema definition, wvrsysconf.xsd conforms to http://www.w3.org/2001/XMLSchema, and is defined from http://www.w3.org/2001/XMLSchema-instance. It contains all the definitions for the system telephony parameters, together with the restrictions that apply to them.
Definition of wvrsysconf.xml

As shown in the example file below, the XML document syntax of wvrsysconf.xml is defined, and constrained by the XML schema definition, wvrSysConf.xsd. The XML namespace is defined as http://www.ibm.com/voice/wvr/config

The root element of the document is wvrSystemConfig, within which are sections for each of the relations in the database. If there are multiple records for a relation then the section can be repeated. The attribute key is used to specify the record number; though this can be omitted for relations that have a single record.

```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<wvrSystemConfig xmlns="http://www.ibm.com/voice/wvr/config"
    xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
    xsi:schemaLocation="http://www.ibm.com/voice/wvr/config wvrSysConf.xsd">
    <!-- Application Server Interface -->
    <asi key="1">
        <!-- Number of 3270 Exec Processes to Spawn -->
        <_3270_num_exec>32</_3270_num_exec>
        <!-- Max Number of Screens Saved by 3270 Exec -->
        <_3270maxscrns>1000</_3270maxscrns>
        <!-- Prompt Volume Ceiling Default (dBm) -->
        <Music2Fgdb>0</Music2Fgdb>
    </asi>
</wvrSystemConfig>
```

The wvrsysconf command

**Purpose**

Allows direct manipulation of the system telephony parameters as an alternative to using the graphical menus.

**Syntax**

To export the parameters for editing:

```
wvrsysconf -e [-f filename] [-u profile [-p password]] [-h]
```

To import the parameters after editing and validating:

```
wvrsysconf -i -f filename [-u profile [-p password]] [-h]
```

**Flags**

- **-e** Specifies export. Schema definition file wvrSysConf.xsd is generated; if a filename is not specified in the -f option, the data file wvrSysConf.xml is also generated.
–i Specifies import. Parameter values are imported from the file that is specified in the –f option.

–f filename
   The name of the file to be used for exporting or importing the telephony parameters.

–u profile
   The name of the administrator profile. If this flag is not specified, a prompt is displayed requesting the profile and associated password (the password is not echoed to the screen).

–p password
   The password associated with the administrator id. If this flag is not specified, a prompt is displayed requesting the password (the password is not echoed to the screen).

–h If this flag is included, only the help screen is displayed.

Exit status

One of the following values is returned (if the –h flag is specified):

0 Successful completion.
1 Failed to open message catalog
2 Password check failed
3 Failed to access database
4 Failed to access database master record
5 Failed on screen IOCTL
6 Help
7 Usage error
8 Import failed
9 Export failed

What next?

Generally, your telephony configuration is now complete. Depending on your requirements, however, you might need to use the instructions in Chapter 5, “Exchange data links and common channel signaling,” on page 91 and Chapter 6, “Advanced system parameter settings,” on page 101, to do further configuration.
When you have restarted WebSphere Voice Response and you are sure that the configuration has worked, make a backup copy of the system configuration (see “Making a backup copy of system parameter values” on page 19).

Your telephony packs are now ready to handle incoming and outgoing calls, but you will also need to set up application profiles, using information in the WebSphere Voice Response for AIX: Managing and Monitoring the System book.
Chapter 5. Exchange data links and common channel signaling

This chapter tells you how to do some additional configuration tasks:

- "Setting the exchange data link parameters"
- "Configuring the CallPath_SigProc signaling process" on page 96
- "Setting extra parameters for ISDN" on page 98

For background information, see "Exchange data link “ on page 28, "Common channel signaling “ on page 28, and "Configuring WebSphere Voice Response to get the called and calling numbers” on page 35.

Setting the exchange data link parameters

If you are using **System Configuration** to define the telephony environment, you must set some parameters in the Exchange Data Link parameter group to define the optional exchange data link. Whether you are using **Pack Configuration** or **System Configuration**, you must set some parameters in the Application Server Interface parameter group.

**Note:** This section applies only if you are planning to use an exchange data link.

When using SMDI, SMSI, or VMS, ensure that the switch to which the exchange data link is connected is set to 7 bits per ASCII character. This does not apply to ACL.

Make sure that getty is not running on the communications port (TTY port) that you are going to use.

For more information, see "Configuring WebSphere Voice Response to get the called and calling numbers” on page 35.

**Procedure**

1. From the Welcome window, select **Configuration —> System Configuration —> Change**.
2. Click **Exchange Data Link**.
3. Set the Exchange Data Link parameters as shown in Table 11.

4. Close the Exchange Data Link window.

Table 11. Exchange data link parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>SMSI, SMDI or VMS Exchange data link</th>
<th>ACL Exchange data link</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDL switch Type</td>
<td>Set this parameter to:</td>
<td>Set this parameter to Siemens.</td>
</tr>
<tr>
<td></td>
<td>• AT&amp;T/Lucent (for an SMSI link)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Nortel (for an SMDI link)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Ericsson (for a VMS link)</td>
<td></td>
</tr>
<tr>
<td>EDL communication port</td>
<td>Set this parameter to /dev/tty$n$, where $n$ is the number of the port used by the exchange data link.</td>
<td>Set this parameter to /dev/mp$qn$, where $n$ is the number of the port used by the exchange data link.</td>
</tr>
<tr>
<td>EDL data rate (bits/sec)</td>
<td>Set this parameter to the data rate recommended by the switch or modem manufacturer.</td>
<td>Not used.</td>
</tr>
<tr>
<td>EDL parity</td>
<td>Set this parameter to the parity recommended by the switch or modem manufacturer.</td>
<td>Not used.</td>
</tr>
</tbody>
</table>
### Table 11. Exchange data link parameters (continued)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>SMSI, SMDI or VMS Exchange data link</th>
<th>ACL Exchange data link</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDL line identifier number length</td>
<td>Change the default value (7) if the switch sends message info line identifiers of more than seven characters. (This is ignored by CallPath_SigProc.)</td>
<td>Not used.</td>
</tr>
<tr>
<td>System number</td>
<td>VMS only: Set this parameter as required.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Called number stripping</td>
<td>Set this parameter to <strong>Yes</strong> to convert the fixed-length numbers received from the switch to the variable-length numbers that are expected by the application. Set this parameter to <strong>No</strong> if the length received from the switch is expected by the application.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Called number length (Minimum)</td>
<td>If <strong>Called Number Stripping</strong> is set to <strong>Yes</strong>, set this parameter to the minimum number of digits in the called number after padding characters are stripped.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Called number character to strip</td>
<td>If <strong>Called Number Stripping</strong> is set to <strong>Yes</strong>, set this to the character the switch uses to pad the called number.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Called number length</td>
<td><strong>SMSI/SMDI only:</strong> Not used. <strong>VMS only:</strong> Set this parameter to the number of digits expected from the switch.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Calling number stripping</td>
<td>Set this parameter to <strong>Yes</strong> to convert the fixed-length numbers received from the switch to the variable-length numbers that are expected by the application. Set this parameter to <strong>No</strong> if the length received from the switch is the same as the length expected by the application.</td>
<td>Not used.</td>
</tr>
</tbody>
</table>
### Table 11. Exchange data link parameters (continued)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>SMSI, SMDI or VMS Exchange data link</th>
<th>ACL Exchange data link</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling number length (Minimum)</td>
<td>If Calling Number Stripping is set to Yes, set this parameter to the minimum number of digits in the calling number after padding characters are stripped.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Calling number character to strip</td>
<td>If Calling Number Stripping is set to Yes, set this to the character the switch uses to pad the calling number.</td>
<td>Not used.</td>
</tr>
<tr>
<td>Calling number length</td>
<td>SMIS/SMDI only: Not used. VMS only: Set this parameter to the number of digits that is expected from the switch.</td>
<td>Not used.</td>
</tr>
<tr>
<td>MWI number length</td>
<td>Set this parameter to the number of digits that is in the MWI number sent to the switch.</td>
<td>Not used.</td>
</tr>
<tr>
<td>MWI number padding</td>
<td>Set this parameter to Yes to pad the MWI number up to a fixed-length before it is sent to the switch. Set this parameter to No if there is no change in the number before it is sent to the switch.</td>
<td>Not used.</td>
</tr>
<tr>
<td>MWI number padding character</td>
<td>Set this parameter to the character to be added to the MWI number before it is sent to the switch.</td>
<td>Not used.</td>
</tr>
</tbody>
</table>

5. Setting Application Server Interface parameters: Open the Application Server Interface parameter group.

6. EDL Call Information After Off Hook: Scroll down to EDL Call Information After Off Hook:
7. Some switches associate call information with a channel only if the channel is active (off hook) before call setup. If this is the condition, set this parameter to Yes.
Other switches require the channel to be inactive (on hook). If this is the condition, set this parameter to No.
8. **EDL Message Info Age Limit**: Scroll down to EDL Message Info Age Limit.

9. Set this parameter to the maximum age of usable information. The unit of measure is seconds.
10. **EDL Message Info Time Out**: Set this parameter to the length of time that WebSphere Voice Response is to wait for information before timing out.

11. **Saving the Parameter Values**: Close the Application Server Interface window and click **File —> Save**.

Although the System Configuration window is still displayed, the system has saved the parameter values.

12. **Close** the System Configuration window.

Now that you have set the Exchange Data Link and Application Server Interface parameters, you have set up your system to use an exchange data link. You must restart the system for the new configuration to take effect. If you are using CallPath_SigProc to get information from the switch using CallPath Server, you must also configure the signaling process (see "Configuring the CallPath_SigProc signaling process").

Ensure you save a copy of the system configuration (see "Making a backup copy of system parameter values" on page 19).

**Note**: When using SMSI, SMDI, or VMS, ensure that the switch to which the exchange data link is connected is set to 7 bits per ASCII character.

---

**Configuring the CallPath_SigProc signaling process**

To use CallPath Server to pass information to and from the switch, you must specify to the signaling process the services you want it to perform.

**Procedure**

1. From the Welcome window, select **Applications —> Custom Servers**.

2. **Opening the custom server**: Open the CallPath_SigProc custom server.

   The system displays the Custom Server window, showing CallPath_SigProc with a Status entry of **Installed**.

3. **Specifying the capabilities to be used**: Select **File —> Properties**.

   The **main() args** pane lists the following entries:
   - `-sn server_name`
   - `-st switch_type`
   - `-sc ANI_DNI`S
   - `-sc FE_HUP`
   - `-sc TRANSFER`
   - `-sc MWI`

4. Alter these parameters to suit your system, as follows:
Specifies the server_name: the name of either the CallPath Server or the CallPath server and can be full server name or TCP/IP address if required.

Specifies the switch_id: the name by which CallPath knows the switch.

Specifies the switch_type: ATTG3i is the only valid value.

Specifies the capabilities that the signaling process is to provide:

- ANI_DNIS
  Specifies that the signaling process is to provide called and calling numbers to WebSphere Voice Response.

- FE_HUP
  Specifies that the signaling process is to provide far end hangup detection capability.

- TRANSFER
  Specifies that WebSphere Voice Response is to use the signaling process for call transfer instead of using the transfer call request signal.

- MWI
  Specifies that the signaling process is to send message waiting indication (MWI) to the switch when necessary.

Starting the signaling process

Start the signaling process in the same way that you start any custom server.

1. File: From the Welcome window, select Operations —> Custom Server Manager.
2. Find the signaling process you want to start.
3. Click NONE under Run Status.
4. Click Start.

Stopping the signaling process

Stop the signaling process in the same way that you stop any custom server.

1. From the Welcome window, select Operations —> Custom Server Manager.
2. Find the signaling process you want to stop.
3. Click WAITING under Run Status.
4. Click Stop.
Setting extra parameters for ISDN

Whether you are using Pack Configuration or System Configuration to define the telephony environment, and you are using ISDN, you might need to set some parameters in the Application Server Interface and ISDN Signaling parameter groups.

Procedure

1. From the Welcome window, select Configuration —> System Configuration —> Change.
2. Click Application Server Interface.
3. Extra Channel Process: When the telephony load is very high, you might see the following message if you are using a common channel signaling protocol such as ISDN:

   No free CHPs could be allocated

   This message indicates that an incoming call was not answered by WebSphere Voice Response because too few channel processes have been configured. By default, the number of channel processes is 10 greater than the number of channels but, with a CCS protocol, this number might need to be increased.

   The Extra Channel Process parameter controls the number of channel processes. If you increase this value by a large amount, you might also need to increase the maximum number of user processes that are configured in AIX, (for details, see the WebSphere Voice Response for AIX: Installation book).
4. Close the Application Server Interface window.
5. Called/Calling Party Numbering: Open the ISDN Signaling parameter group.

6. Set the L4 - Called/Calling Party Numbering Type and L4 - Called/Calling Party Numbering Plan parameters according to the numbering used in your network.

   If you do not know the numbering Type and Plan that are specific to your network, you can try setting the parameters to Unknown. However, if you continue to have problems, see your network documentation for the numbering type and plan that is used in your network.

7. Close the ISDN Signaling window
8. **Saving the Parameter Values:** Save the new values. Although the System Configuration window is still displayed, the system has saved the parameter values.

9. **Close** the System Configuration window.

You have now completed configuration for ISDN. You must shut down and restart WebSphere Voice Response for the changes to take effect. For guidance on enabling the trunk and channels and monitoring the activity on the ISDN lines, see the *WebSphere Voice Response for AIX: Managing and Monitoring the System* book.

Ensure that you save a copy of the system configuration (see “Making a backup copy of system parameter values” on page 19).
Chapter 6. Advanced system parameter settings

This chapter describes how to use the following types of system parameter:

- “Setting call progress tone parameters for outbound dialing”
- “Setting parameters for hangup tone detection” on page 112
- “Setting parameters for voice interrupt detection” on page 117
- “Setting line code and framing mode parameters” on page 119
- “Setting parameters for voice-data compression” on page 121
- “Setting parameters for redial limitation” on page 122

Setting call progress tone parameters for outbound dialing

Call progress tones are primarily used for outbound dialing and call transfer. Tracking the tones allows WebSphere Voice Response to return the status of a call to the state table that attempted to make or transfer it. The state table can then take suitable action in response. For example, if a busy tone is detected, the state table can terminate the call and try again later.

Note: This section applies only to Channel associated signaling (CAS).

WebSphere Voice Response can detect the following types of tone:

- **Single tones** (one-frequency tones)
- **Tritones** (three sequential tones at different frequencies)
- **Dual tones** (two simultaneous tones at different frequencies)

WebSphere Voice Response cannot detect dual sequential tones (sequential tones that, in turn, consist of dual tones).

To support tone detection, parameters must be configured correctly to match those sent by the national telephone network or by the local PBX.

After you have used Pack Configuration to define your telephony environment, the system can recognize standard call progress tones that are sent by your national telephone network. You can use the Pack Configuration window to view the call progress tone settings. If you need to change these, for example, to configure the system to recognize tones sent by a specific PBX, ask your IBM representative to use System Configuration to reset the values. To do this, the IBM representative needs a list of the nominal values and valid ranges for tones provided by the switch.
Using call progress tone detection for outbound calls

This section outlines how call progress tones are used for outbound dialing. For specific information about using call progress tones in a state table, see the Dial, MakeCall, TransferCall, and ReconnectCall actions in the WebSphere Voice Response for AIX: Application Development using State Tables book.

A telephone call from one person to another typically proceeds as follows:
1. First, the caller takes the handset off the hook and listens for a dial tone.
2. Having confirmed the presence of dial tone, the caller proceeds to dial.
3. The caller then listens for an audible ring tone.
4. When the called party answers, the caller detects a voice and knows that the call has been answered. The call setup is now complete.

Of course, other possible results can occur during call setup, such as hearing a busy tone instead of an audible ring. A caller would hang up and try again later.

WebSphere Voice Response makes outbound calls in much the same way.

The application developer can program the state table to control the decisions during call setup:
1. When a Dial, MakeCall, ReconnectCall, or TransferCall action goes off-hook, the call progress tone function is activated to search for a dial tone. If a dial tone is not detected, several possibilities exist, including hangup. For more information, see the Dial, MakeCall, TransferCall, and ReconnectCall actions in the WebSphere Voice Response for AIX: Application Development using State Tables book.
2. If a dial tone is detected, dialing follows.
3. After the number is dialed, the call progress tone function is activated again to search for an audible ring tone.
4. Several possible results can be returned by the call progress tone function:
   - If the destination phone rings, a positive audible ring result is returned.
   - If the phone is engaged, a busy (line) result is returned.
   - If the phone is answered, a voice energy presence result is returned.

In summary, WebSphere Voice Response detects a specific type of tone at each step during call setup. The returned value indicates the result of the tone search, which in turn indicates the status of a call.

A WebSphere Voice Response application can receive an unexpected tone; that is, a tone that should not occur at a particular point in the dialing sequence. For example, a dial tone should not occur after a ring tone. If such an
unexpected tone is received, WebSphere Voice Response then returns the Tone Identifier and the Tone Type to the application (see "How call progress tones are identified" on page 104).

The application can then perform some action, related to the unexpected tone it receives.

**Call progress tone detection performance specifications**

Table 12 on page 104 shows the performance specification met by WebSphere Voice Response call progress tone detection. However, note that the performance is degraded in noisy telephony environments.

The call progress tones detector that WebSphere Voice Response uses can detect a single tone, dual tones, or voice energy. The tones must meet the following criteria:

- A tone must be less than 2000 Hz.
- The sum of the tones must be less than 1500 Hz. (If the sum of the frequencies is greater than 1500 Hz, the detector reports the average frequency. The same is true if the two tones are close in frequency; for example, 400 Hz and 430 Hz.)

The detector evaluates the ratio of peak energy to average energy. For a single tone, the ratio is 1:1; for a dual tone, the ratio is 2:1; and for voice energy, the ratio is greater than or equal to 5:1. The audio energy is sampled every 20 ms.

The detector uses "curve fitting" to identify frequencies and energy levels. For example, if two tones are detected, the detector attempts to fit a curve for two tones. If the sum of the frequencies is greater than 1500 Hz, the detector reports the average frequency. With voice energy, the detector reports the amount of time the energy is greater than the value set for the Constant Energy Minimum system parameter.

Continuous tones are detected using the detector’s “immediate mode”, which samples the tones every 100 ms instead of every 20 ms. Only frequencies and energy levels are reported.

**Attention**

In evaluating tones, WebSphere Voice Response searches for the first appearance of the tone in the Call Progress Tones parameter group, beginning with Tone Id 1 (see "How call progress tones are identified" on page 104). For this reason, it is very important that the range that is specified by minimum and maximum values for one frequency does not overlap the range that is specified by the values for another frequency with a different Tone Id number.
Table 12. Call progress tone detection performance specifications

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Tone Type</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Single</td>
</tr>
<tr>
<td><strong>Level (dBm)</strong> (+ or - 2.5 dB)</td>
<td>+6 to -43</td>
</tr>
<tr>
<td><strong>Frequency (Hz)</strong> (+ or - 2.5%)</td>
<td>200 to 2000</td>
</tr>
<tr>
<td><strong>Cadence</strong></td>
<td>Continuous or cadenced</td>
</tr>
</tbody>
</table>

**Note:**
1. -10 to -34 if the frequency difference is 30 to 50 Hz; -10 to -43 if the frequency difference is greater than 50 Hz. A reduction in dynamic range occurs when the frequency difference for the two tones is less than 50 Hz. An example is U.S. dial tone, which consists of 440 Hz and 480 Hz components.
2. If the sum of the frequencies is greater than 1500 Hz, the detector reports the average frequency. The same is true if the two tones are close in frequency, for example, 400 Hz and 430 Hz.
3. The minimum on and off periods are 60 ms. The maximum on and off periods are 5 seconds.

**How call progress tones are identified**

Up to 25 different tones can be defined for a WebSphere Voice Response system. Each tone is defined by a set of parameters in the Call Progress Tone parameter group. Groups of tone definitions are supplied, to define the tones used in various countries. Pack Configuration sets the tone definitions correctly for each country but, if you need to change them (for example, to detect tones from a specific PBX), follow the procedure in “Redefining call progress tones” on page 109.

Each tone in the tone group has a **Tone Identifier**, which is a sequential identifier that is used by state table actions and in the call progress tone parameter groups to identify the tone. Each call progress tone group includes **Tone Id 1** through **Tone Id 25**.

In addition to the Tone Identifier, each tone has a **Tone Label** and **Tone Type**, which are values assigned to parameters in the tone definition (see “How call progress tones are defined” on page 105).
The **Tone Group** parameter in the Channel Group parameter group specifies which group of tone definitions is to be used on channels in each channel group. This enables you, for example, to detect national network tones on some channels and PBX tones on other channels.

**How call progress tones are defined**

Each tone is defined by the following parameters:

- **Tone Type**
- **Tone Label**
- Frequency (up to three frequency components can be specified)
- Level (of up to three frequency components)
- Cadence (of up to three frequency components)

**Tone Type**

The call progress tones and the tone types vary from country to country. However, four basic types, in addition to “voice activity detection”, exist in all countries. They are:

- **Ring** An audible ringing tone.
- **Dial tone** An audible dialing tone.
- **Busy** An audible line busy (engaged) tone.
- **Network Busy** A tone that is generated when all central office circuits are in use. Also known as *equipment busy*.

**Other** The availability of other call progress tones in a specific country or region is dependent upon the telephone system of that country or region.

**Note:** Some of these tones are not already defined to WebSphere Voice Response. To enable WebSphere Voice Response to use these, you must follow the procedure described in “Redefining call progress tones” on page 109.

The tone type is specified by the Tone Type parameter in the call progress tone definition.

**Tone Label**

The tone label is a description of the tone that you specify by using the **Tone Label** parameter in the call progress tone definition.

**Frequency**

The frequencies are defined in Hertz (Hz). Up to three frequencies can be defined for each tone. A **Frequency Maximum** and a **Frequency Minimum** parameter value for each frequency component defines the qualifying range.
The minimum value for tone frequencies is nominal frequency - 9% and the maximum value for tone frequencies is nominal frequency +9%.

Double interrupted tones, for example, 400 ms on, 200 ms off, 400 ms on, 2000 ms off, must be split into two parts. For an example of how to do this, see Ring Part 1 Tone and Ring Part 2 Tone in the tone parameter table for the United Kingdom.

**Level**

The levels are defined in decibels (dBm). A **Level Maximum** and a **Level Minimum** parameter value for each frequency component defines the qualifying range.

The minimum level for each tone is set to 6 dBm lower than the expected minimum value. Set the maximum level to 0 dBm.

**Cadence**

The cadences are defined in terms of Time On and Time Off. A **Time On Maximum** and a **Time On Minimum** parameter value for each frequency component defines how long a signal of the specified frequency and level must be to qualify as a valid component of the tone. A **Time Off Maximum** and a **Time Off Minimum** parameter value for each frequency component defines how long a period must elapse between signals. No Time parameters are specified for a continuous tone.

The values for times are set in 20 ms increments, rounded to the next lowest for minimum value and to the next highest for maximum value. The minimum value for a tone cadence is specified as nominal time (on or off) -15% and the maximum value for a tone cadence is specified as nominal time (on or off) +15%.

**Example: How to work out maximum and minimum values from nominal values**

To define a dual tone specified by the nominal values shown in Table 13, perform the calculations shown in Example call progress tone calculations to work out the maximum and minimum values:

<table>
<thead>
<tr>
<th>Frequency 1</th>
<th>Frequency</th>
<th>Level</th>
<th>Time On</th>
<th>Time Off</th>
</tr>
</thead>
<tbody>
<tr>
<td>300 Hz</td>
<td>-18 dBm</td>
<td>250 ms</td>
<td>250 ms</td>
<td></td>
</tr>
</tbody>
</table>
Table 13. Example call progress tone nominal values (continued).

This table describes for a dual tone, the nominal values of the frequency in Hz, level in dBm, time on and time off in ms for Frequency 1 and Frequency 2, where Frequency 1 is the lower of the two frequencies.

<table>
<thead>
<tr>
<th>Frequency 2</th>
<th>Frequency</th>
<th>Level</th>
<th>Time On</th>
<th>Time Off</th>
</tr>
</thead>
<tbody>
<tr>
<td>480 Hz</td>
<td>-18 dBm</td>
<td>250 ms</td>
<td>250 ms</td>
<td></td>
</tr>
</tbody>
</table>

Notes:
1. For a dual tone, Frequency 1 must be the lower of the two frequencies. For tritones, Frequency 1 must be the first tone, Frequency 2, must be the second tone, and Frequency 3 must be the third tone in the sequence.
2. Use 9% as a guideline only in calculating minimum and maximum frequency values. With smaller absolute numbers, a larger percentage should be used. Also, you need to ensure that the range specified by minimum and maximum values for one frequency do not overlap with that specified by the values for another frequency.
3. All time values must be rounded to the nearest 20 ms.

Example call progress tone calculations

<table>
<thead>
<tr>
<th>Frequency 1</th>
<th>Frequency 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td>= 300 - 9%</td>
<td>= 300 + 9%</td>
</tr>
<tr>
<td>= 300 - 27</td>
<td>= 300 + 27</td>
</tr>
<tr>
<td>= 273 Hz</td>
<td>= 327 Hz</td>
</tr>
<tr>
<td>Level 1</td>
<td>Level 2</td>
</tr>
<tr>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td>= -18 + (-6)</td>
<td>= 0 dBm</td>
</tr>
<tr>
<td>= -24 dBm</td>
<td></td>
</tr>
<tr>
<td>Time On 1</td>
<td>Time On 2</td>
</tr>
<tr>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td>= 250 - 15%</td>
<td>= 250 + 15%</td>
</tr>
<tr>
<td>= 250 - 38</td>
<td>= 250 + 38</td>
</tr>
<tr>
<td>= 212</td>
<td>= 288</td>
</tr>
<tr>
<td>= 200 ms</td>
<td>= 300 ms</td>
</tr>
<tr>
<td>Time Off 1</td>
<td>Time Off 2</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Recognizing double rings

Two ring tones should be defined when a double ring tone needs to be detected. A double ring tone is one in which the ring sequence is: ring...short pause...ring...long pause. The sequence, however, might not start at the beginning, which means that what WebSphere Voice Response detects is: ring...long pause...ring...short pause. To detect either sequence, then, you must define two tones, each with the same frequencies and levels, but with different cadences. One is defined as the first portion of the ring (ring...short pause), and the other is defined as the second portion (ring...long pause). This enables faster, more efficient detection of double ring tones.

Displaying call progress tone values

This method is the easiest way of displaying the call progress tone values in use for each channel group. This method is accessible to anyone who has access to the Pack Configuration window.

Procedure

1. From the Welcome window, select Configuration —> Pack Configuration.
2. Click the Direction & Tones button next to the channel group in which you are interested.
   The system displays the Call Direction window.
3. Click the Display Call Progress Tones button.
   The system displays the Call Progress Tones window:
For an explanation of each column heading, click the Help button.

4. When you have finished viewing, click Cancel.
5. Click File —> Close.

To change the values, see “Redefining call progress tones”

Redefining call progress tones

If you have specified your country or region by using Pack Configuration, the call progress tones are set up as correct for the national public switched telephone network. However, some of the call progress tones that are issued by your switch or channel bank might not be the same as those issued by the network. In such conditions, the tone parameter values must be redefined.

Procedure

In outline, the procedure is as follows: copy the appropriate country- or region-specific table to one of the numbered tone groups, edit that tone group so that the frequency and pitch of each tone matches the tones provided by your switch, and then update the Tone Group parameter in the relevant
channel groups so that WebSphere Voice Response will use your customized tone group instead of the default table for your country or region.

1. From the Welcome window, select Configuration —> System Configuration —> Change.
2. Click Call Progress Tones.
3. Accessing the Call Progress Tones parameter group:

   ![Call Progress Tones Table]

   **Note:** In some countries, the telecommunications authorities have ruled that call progress tone configuration must be performed by an IBM representative. For this purpose, a special password-protected administrator profile, field, is provided, which gives access to field-level parameters, including the Call Progress Tone groups.

   If the Call Progress Tones button is not displayed in the System Configuration window, you must log off and log on to WebSphere Voice Response again, this time using the user ID and password for the field administrator profile.

   Selecting and copying a template
4. Select the country or region or switch whose tone data you want to customize.
5. Follow the instructions in “Copying parameter values” on page 18.
6. **Changing the parameter values:** Click File —> Open to display the newly-copied call progress tone table.
   The system displays a window which contains 25 tone buttons, each representing one specific tone.
7. Click a button that represents a tone that you wish to update. Click File —> Open to display details of the tone.

8. Click a button that represents a specific tone parameter that you need to change. Click File —> Open to display the tone parameter window.
9. Change the contents of the New Value field as necessary. Ensure that the value you enter here is inside the minimum and maximum permitted values (as displayed in the parameter window).

10. Click OK.

11. The system closes the tone parameter window; your newly-entered value is now shown next to the related button.

12. Repeat Steps 8 on page 111 for any other parameters that require updating.

13. Click File —> Close.

14. Repeat Steps 7 on page 111 for any other tones that require updating.

15. Click File —> Close.

16. Click File —> Close.

17. Saving the new information: Click File —> Save.

18. Click File —> Close.

19. You are returned to the Welcome window.

The new tone group is now complete. To use it, you must now specify this tone group in the Tone Group parameter for all channel groups that need to use call progress tones. Note that if Pack Configuration has been used to set the country or region for the system, Channel Group 1 has been set up to use Tone Group 1 and Channel Group 2 has been set up to use Tone Group 2.

If you do not want to wait until WebSphere Voice Response is next restarted, you must use the System Monitor window to disable, then enable, the associated trunks.

---

**Setting parameters for hangup tone detection**

To operate most effectively, a voice application needs to be able to detect, as soon as possible, when the caller, or called party, hangs up without completing the transaction. Otherwise, you might be paying for unnecessarily long calls. Some switches issue a constant tone to indicate that the far end has hung up. Other switches, particularly those using loop start protocols, issue a cadenced tone to indicate far-end hang up. Providing the Signaling Type parameters are configured correctly, WebSphere Voice Response can detect these tones, and pass the information on to the voice application so that it can end the call.

**Constant tone detection**

Constant hangup tone detection is relatively simple. It is controlled by the following parameters in the Signaling Type parameter group:
The Hangup Detection parameter must be set to **Constant Energy Detection**. The Constant Energy Minimum and Constant Energy Maximum parameters define the energy band inside which a constant tone is assumed to be the hangup tone. If the tone you get from your switch is not being detected with the default values, you need to reset these parameters.

**Cadenced tone detection**

With cadenced tones (also known as “interrupted” tones), detection is a little more difficult. It is controlled by the following parameters in the Signaling Type parameter group:

- Cadence Energy Maximum (dBm) (page 217)
- Cadence Energy Minimum (dBm) (page 218)
- Cadence Off Time Maximum (ms) (page 219)
- Cadence Off Time Minimum (ms) (page 220)
- Cadence On Time Maximum (ms) (page 221)
- Cadence On Time Minimum (ms) (page 222)
- Cadence Silence Maximum (dBm) (page 223)
- Hang Up Detection (page 336)

The Hangup Detection parameter must be set to **Cadence Energy Detection**.

*Cadenced* means that the hangup signal is intermittent. It consists of a series of signals (known as on phases) separated by silence (known as off phases). WebSphere Voice Response uses a cadenced tone detector to monitor the energy levels of the on and off phases and how long each phase lasts.

The cadenced tone detector runs constantly except when the channel is dialing, and when DTMF is being received. The start of dialing or the arrival of a DTMF tone causes the cadenced tone detector to reset.

WebSphere Voice Response waits for two on phases and two off phases (on-off-on-off) to pass before confirming that the signal is indicating hangup.

**Energy Levels**

The energy level of the on pulses must be within a range of values known as the high band and the off pulses must be within a range of values known as the low band. These bands are controlled by the parameters Cadence Energy...
Maximum, Cadence Energy Minimum and Cadence Silence Maximum as shown in 114. The energy levels of the electrical signals are measured in dBm. The frequency (pitch) of the tone is not checked.

The detector checks that the dBm level of all the on pulses are between the Cadence Energy Minimum and Cadence Energy Maximum values. The on pulse is allowed to settle or rise to a level within the configured band of allowable dBm range until it stabilizes; after that, it must remain relatively constant (±1dBm) until the pulse ends.

Timing

The detector also monitors the length of time taken for each on pulse and the length of time taken for each off pulse. The time taken for an on pulse must always be within a range of values controlled by the parameters Cadence On Time Minimum and Cadence On Time Maximum. Similarly, the time taken for
an off pulse must always be within a range of values controlled by the parameters Cadence Off Time Minimum and Cadence Off Time Maximum. This is shown in [115].

The detector reports the hangup when it detects the rising edge of the third on-pulse.

**Settle time**

[116] shows a signal that starts at a low dBm level, and rises to a high dBm level not smoothly, but in steps. Part of this stepping is caused by the electrical properties of the line and hardware used to generate the pulse, and...
part is caused because energy levels are being reported every 20 ms.

The signal must settle down to a steady (or nearly steady) energy level before the settle time expires. After the settle time has expired, the dBm level of the on-pulse is checked every 20 ms to ensure it differs from the settled dBm level by no more than 1 dBm. Failure to do this causes the detector to wait for the next valid off period, then start searching for the next on-period again. This exceptions to this behavior are when glitches are detected (see [17]).

The settle time is defined as 60 ms constant.

During the settle time, the dBm level is not monitored except that, if it leaves the High Band, the detector resets and waits for the next valid off time before restarting.

During the off period in the cadenced sequence, the dBm level is monitored only to ensure that it does not rise above the threshold specified by Cadence Silence Maximum. If it does, the detector resets and starts looking for a valid silence before restarting.

If the energy level ever increases beyond the level specified by Cadence Energy Maximum, although that increase might be inside 1 dBm of the settled dBm level, the detector resets and waits for the next valid off period before restarting.
This settle time applies to all rising edges; that is, at the start of all on-phases of the cadenced sequence.

Glitch-detection is not active during the settle time.

**Glitches**

A *glitch* (also known as a *spike*) is caused by a brief variation in the energy level of the sample, during either the on-phase or off-phase of the cadence sequence. Such glitches are generally ignored by the detector, so two consecutive on-phases and off-phases, followed by a third rising edge, can still be detected, although the line might be noisy.

The detector considers any deviation from the settled on-time dBm level to be a glitch, providing it lasts for less than 40 ms. If the dBm level does not return to the settled level for more than 40 ms, it is considered to be a nonvalid on-signal, and the detector resets and starts again.

The detector can ignore a small number of glitches in any single on-state or off-state. More than this in any single state causes the detector to restart.

Glitches are not detectable during the settle time.

---

**Setting parameters for voice interrupt detection**

Voice interrupt detection allows a caller to interrupt the playing of a prompt or a voice segment by speaking. Voice interrupt detection is controlled using three system parameters and four system variables. The system parameters specify default values for the whole system. These three parameters can be overridden for an individual application by the use of system variables. The other system variable turns voice interrupt detection on or off. This action cannot be done system-wide.

Table 14 shows the system variables and their equivalent system parameters:

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>System Variable</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Interrupt Detection Level</td>
<td>System : Voice Interrupt Detection Level (SV218)</td>
</tr>
</tbody>
</table>
Table 14. System parameters and variables used by voice interrupt detection (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>System Variable</th>
</tr>
</thead>
</table>

The value of Voice Interrupt Detection Level specifies the minimum energy level that the voice interrupt detector considers to be an interrupt. That is, any noises below that level do not count as interrupts.

Be careful when setting the Voice Interrupt Detection Level. If it is too high, the caller cannot interrupt prompts by speaking normally or shouting. If the value is too low, the echo\(^4\) from a prompt that is being played, combined with the background noise level from the caller’s telephone, might interrupt prompts unintentionally. If a very loud echo occurs on a particular line, the level should be increased.

The value of Voice Interrupt Detection On Time specifies the minimum length of time for which the audio signal must remain above the minimum energy level. That is, a very short sound does not count as an interrupt.

\(^4\) Echo can be generated by any connectors or switches that have analog circuits. Echo can come from local equipment, the network provider, or the caller’s equipment.

After detecting a sound that qualifies as an interrupt (it is above the minimum energy level and longer than the minimum on time) the detector

\[\text{Figure 13. WebSphere Voice Response detects a voice interrupt}\]
must ensure that the caller has finished speaking the word before it starts the next state table action. This is because the caller might have spoken a multi-syllable word as the interrupt; for example, “cancel”. If the detector stops when it has heard the first syllable, the second syllable might be taken as another word and sent to a speech recognizer. (A tone played between the play and record actions in the state table ensures that the caller's interrupt completes before the record action begins.) So, the Voice Interrupt Detection Off Time value is used to ensure that a period of silence occurs after the interrupt-word has been spoken. This also ensures that a continuous sound picked up by the caller's telephone is not assumed to be an interrupt.

Default values exist for all of these parameters and variables, which work in most circumstances. Change these values only if you need to; for example, if levels are set too high or too low for the environments in which your applications are used. See the WebSphere Voice Response for AIX: Problem Determination book for examples of problems you might fine using voice interrupt detection and suggested solutions.

More information about voice interrupt detection is given in the WebSphere Voice Response for AIX: Designing and Managing State Table Applications book.

### Setting line code and framing mode parameters

Whether you have used System Configuration or Pack Configuration to configure the telephony environment, you might need to set suitable values for system parameters in the Trunk Interface parameter group.

**Note:** In some countries, the telecommunications authorities have ruled that parameters that affect the connection with the telephone network must be performed by an IBM representative. For this purpose, a special password-protected administrator profile, field, is provided, which gives access to field-level parameters, including the Line Code and Framing Mode parameters.

#### E1 line code and framing mode

For E1 trunks, the line coding is always HDB3 (high-density bipolar of order 3). This is similar to B8ZS. No system parameter needs to be set.

The E1 Framing Mode parameter (in the Trunk Interface parameter group) should be set to the correct value, as shown in Table 15 on page 120. The possible values are:

**Double**

Double Frame has two concatenated frames (this is the default.)
CRC  CRC, or multiframe, has 16 concatenated frames. This type of framing also has cyclic redundancy checking and is recommended for ISDN connections.

Table 15. E1 Framing mode values for different signaling protocols

<table>
<thead>
<tr>
<th>Signaling Mode</th>
<th>Protocol</th>
<th>Country or region</th>
<th>E1 Framing Mode</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel Associated Signaling (CAS)</td>
<td>All</td>
<td>All</td>
<td>Double</td>
</tr>
<tr>
<td>Common Channel Signaling (CCS)</td>
<td>ISDN</td>
<td>All</td>
<td>CRC</td>
</tr>
<tr>
<td></td>
<td>Other</td>
<td>All</td>
<td>Double</td>
</tr>
</tbody>
</table>

T1 line code, framing mode, and framing format

For T1 trunks, the T1 Line Code system parameter (in the Trunk Interface parameter group) should be set to the correct value, as shown in Table 16 on page 121. The possible values are:

AMI   Alternate Mark Inversion with Zero-Code Suppression (known as AMI-ZCS). (This is the default.) This does not allow "clear channel", that is 64-kilobit per second operation, whether or not when Robbed-Bit Signaling is used with channel associated signaling (CAS).

B8ZS  Bipolar with 8-Zero Substitution. This is required for 64-kilobit per second channels like ISDN.

The T1 Framing Mode system parameter (in the Trunk Interface parameter group) should be set to the correct value, as shown in Table 16 on page 121. The possible values are:

D3/D4 Superframe with 12 concatenated frames. (This is the default.)

ESF   Extended Superframe with 24 concatenated frames. This type of framing also has cyclic redundancy checking and a maintenance channel, and is used for common channel signaling (CCS).

The T1 CAS Signaling Format system parameter (in the Signaling Type parameter group) should be set to the correct value, as shown in Table 16 on page 121. This specifies the T1 bit-robbed CAS Signaling format (as defined in TIA/EIA-464-B) to be used. The possible values are:

2-bit AB (SF) Only the AB bits are used for signaling; C and D are ignored. (This is the default.)
4-bit ABCD (ESF)

All the ABCD bits are used for signaling. The C and D bits are normally set to the values of the A and B bits, but this depends on the protocol (for more information, see TIA/EIA-464-B).

If the T1 framing mode on the trunk is set to D4, choose 2-bit format. If the T1 framing mode on the trunk is ESF, choose 4-bit format.

Because the signaling format is defined by channel group, you must ensure that channel groups that span multiple trunks all have the trunks set to use either D4 or ESF; otherwise some signaling problems might occur.

Table 16. T1 line code and framing mode values for different signaling protocols

<table>
<thead>
<tr>
<th>Signaling Mode</th>
<th>Protocol</th>
<th>T1 Line Code</th>
<th>T1 Framing Mode</th>
<th>T1 Framing Format</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel Associated Signaling (CAS)</td>
<td>All</td>
<td>B8ZS or AMI</td>
<td>D3/D4 or ESF</td>
<td>2-bit (SF) or 4-bit (ESF)</td>
</tr>
<tr>
<td>Common Channel Signaling (CCS)</td>
<td>ISDN</td>
<td>B8ZS</td>
<td>ESF</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Other</td>
<td>B8ZS</td>
<td>D3/D4 or ESF</td>
<td></td>
</tr>
</tbody>
</table>

Setting parameters for voice-data compression

In versions of DirectTalk before Version 2 Release 2, audio names and user greetings were always compressed before they were stored. Now you can specify that all new audio names recorded using state table actions are compressed, or that they are all not compressed. You specify this by using a WebSphere Voice Response system parameter named "Audio Name CompressionType" on page 212 (see "Audio Name CompressionType" on page 212). You can do the same for user greetings by using the new "User Greeting Compression Type" on page 562 system parameter (see "User Greeting Compression Type" on page 562).

When an audio name or user greeting is compressed, it occupies approximately 5 times less disk space than it would if it were uncompressed. On playback, some of the original audio signal is lost, and in some installations, users might notice a reduction in quality. This can occur if the voice data is recorded in a noisy environment.
An audio name or user greeting that is uncompressed uses all the original signal when it is played back, so no reduction in quality occurs. However, uncompressed voice data requires more disk space to store it, and more memory to process it than compressed data does.

**Setting parameters for redial limitation**

The redial limitation facility allows you to configure WebSphere Voice Response to prevent further calls to number after it has been dialed unsuccessfully a certain number of times in a certain time period. You can control this facility using the following system parameters:

- ISDN - Redial Limitation (page 350)
- Redial Limitation - Failed List Capacity (page 445)
- Redial Limitation - Maximum Consecutive Failures (page 446)
- Redial Limitation - Significant Digits (page 447)
- Redial Limitation - Timeout (page 448)

These parameters apply to ISDN only. You cannot use this facility if you are using other CCS protocols or CAS signaling channels.

To enable redial limitation, set the ISDN - Redial Limitation parameter to yes. This parameter is automatically set to yes in Japan.

When redial limitation is enabled, the destination and time of all unsuccessful outbound calls are logged on a failed call list. By default this list can hold up to 1000 numbers. You can increase or decrease the maximum length of the list using the Redial Limitation - Failed List Capacity parameter.

Numbers on the list can only be dialed unsuccessfully a further two times before WebSphere Voice Response refuses any further requests to place a call to that destination. The number of unsuccessful attempts allowed before the number is temporarily blocked can be changed with field authority only, using the Redial Limitation - Maximum Consecutive Failures parameter.

A number is removed from the list after a successful call to that destination or after three minutes have elapsed. This time period can be changed with field authority only, using the Redial Limitation - Timeout parameter. When a number is removed from the list WebSphere Voice Response will no longer refuse calls to that destination.

By default, WebSphere Voice Response logs the entire telephone number to the failed calls list. You can configure WebSphere Voice Response to log only the last n digits of the number, using the Redial Limitation - Significant Digits parameter. This allows area codes to be removed from the logged number. For example, set Redial Limitation - Significant Digits to 6 to store the last 6 digits
of the number. WebSphere Voice Response will then assume that any number dialed with the same 6 final digits as a number on the list is the same number.
Chapter 7. The 3270 host connection

The WebSphere Voice Response 3270 option gives WebSphere Voice Response the ability to communicate with a 3270 host by emulating a 3270 terminal. Terminal emulation enables voice application developers to capture screens and create 3270 servers, and it also enables the 3270 servers to communicate with an application on the 3270 host.

Before WebSphere Voice Response can communicate with the 3270 host, however, the system must be configured to include terminal emulation. In addition, a defined 3270 session is needed to start 3270 terminal emulation.

1. **Configuring the 3270 host connection**
2. **Introducing 3270 session configuration** on page 127
3. **Configuring a 3270 session for screen capture** on page 128
4. **Accommodating new 3270 servers** on page 130
5. **Updating the configuration after changing the hostname** on page 131

Configuring the 3270 host connection

Use this procedure to configure the 3270 host connection when WebSphere Voice Response includes the 3270 option.

When the 3270 option is included, the default mode is real mode. If desired, you can change this to virtual mode during the configuration procedure. The 3270 parameters are not dynamic. After you have assigned values to these parameters, you must stop WebSphere Voice Response and start it again before the new values take effect.

To complete this procedure, you work with parameters in the Application Server Interface parameter group. If a default value is suitable, you do not have to change it. “Application server interface parameter group” on page 180 lists the default values for the parameters in this group.

Procedure

1. From the Welcome window, select Configuration —> System Configuration —> Change.
2. Click Application Server Interface.
3. Defining the number of 3270 processes: Click Number of 3270 Exec Processes to Spawn.

4. Type in the number of processes you want WebSphere Voice Response to start to handle all requests for 3270 emulation.

5. Click OK.

6. Limiting the number of screens saved: Click Max Number of Screens Saved by 3270 Exec.

7. Type in the number of screen definitions you want WebSphere Voice Response to be able to store in the database.

8. Click OK.

9. Changing the installation mode: To operate in virtual mode, scroll down to 3270 Mode, click it, then click the button that is next to Virtual Mode.

10. Select OK.

11. Saving the configuration: Close the Application Server Interface window and click File —> Save.

12. Close the System Configuration window.

13. The system displays the Welcome window.

14. Activating the new values: Click Operations—> Immediate Shutdown or Quiesce Shutdown.

15. The system prompts you to verify.

16. Click OK.

The next time you start WebSphere Voice Response, the system is configured for 3270 emulation.
3270 session configuration involves associating each 3270 server with one or more 3270 sessions. The 3270 servers can then be used to get data from and send data to 3270 host applications, using the 3270 data stream. (For more information about 3270 servers and how they work, see the *WebSphere Voice Response for AIX: 3270 Servers* book.)

**How many sessions?**

WebSphere Voice Response starts the number of processes that is defined by the Number of 3270 Exec Processes to Spawn system parameter in the Application Server Interface parameters group. This value should be greater than or equal to the number of LU2 profiles defined, up to a maximum of 254. If WebSphere Voice Response is to use more than one database host, some of the sessions communicate with one host and some with another.

Because each session can be used by only one server, one way of estimating how many sessions to configure is to estimate how many telephone calls might require sessions at the same time. Another way of estimating how many sessions to configure is to think about the number of channels for which WebSphere Voice Response is configured. Each channel can use up to three sessions at the same time. The more channels your system includes, the more sessions you need. If sessions turn out to be superfluous, they can be removed. They can also be reconfigured for use by a different server at any time.

Because voice application developers use terminal emulation to capture 3270 screens and create servers, always configure at least one additional session for use by the screen capture program.

**When should I configure sessions?**

Sessions can be configured when you need them, or all at the same time. Because you cannot add or delete sessions dynamically, you might want to configure many sessions at first, assigning them to a "dummy" server. You can edit the session configuration after voice application developers have written the 3270 servers that will actually use the sessions.

When you configure the system to allow the 3270 screen capture program to use it, configure at least one session for a 3270 server. You can configure other sessions...
sessions for 3270 servers at another time. You can also use this session to ensure the network definitions are correct and that 3270 emulation is working properly. You cannot invoke 3270 emulation without at least one configured session.

**Before you start**

Before you configure any sessions, ensure that the data communications network includes WebSphere Voice Response, and that all the necessary profiles and definitions have been created and stored on the pSeries computer. The *WebSphere Voice Response for AIX: Installation* book describes how to create profiles and start both Communication Server and all the defined attachments.

To access **3270 Session Configuration** from the **Configuration** menu, complete the procedure that is given in Chapter 7, “The 3270 host connection,” on page 125.

---

### Configuring a 3270 session for screen capture

To configure a session for screen capture, first define a dummy server. (When you configure a session for use by a 3270 server, that server is usually already defined by the voice application developers.) Assign the dummy server to one (or more) of the available sessions. You can configure multiple sessions now, assigning them to the dummy server, then reconfigure them later, and assign them to 3270 servers. “Reconfiguring 3270 sessions” on page 131 describes how to reconfigure sessions.

Server definitions include a session allocation method. If your site has installed a CallPath call processing product, you can specify that a server use a specific session that is tied to a particular telephone number. (When sessions are allocated by telephone number and the session needed by a server is in use, the server must wait until the session is free.) When a server uses the session that is tied to a telephone number and the telephone call is transferred to an agent, CallPath can transfer the session to the agent at the same time.

### Defining a dummy 3270 server

Use this procedure to define a dummy server to configure a session for 3270 screen capture.

1. From the Welcome window, select **Applications** | **3270 Servers**.
2. **Defining a server:** Click **Server** | **New**.
   - The system displays the 3270 Server window.
3. Type in a description of the server. For example, you may want to describe it as “Dummy server to configure emulation session.”
4. Make sure the session allocation method is **First Available**.
5. **Naming the server**: Click File —> Save As...
   
   The system prompts you for a name.

6. Type in a name for the server. The name cannot include blanks, but you can use underscores.

7. Click **OK**.

   Although the 3270 Server window is still displayed, the system has stored the server definition. The name of the server is displayed in the **3270 Server** field. When you close the window, the system displays the 3270 Servers window that lists the new server.

### Configuring a 3270 session

Use this procedure to configure a 3270 terminal session for use by the screen capture program. To define the dummy server to assign to this session, use the procedure that is given in “Defining a dummy 3270 server” on page 128.

This method of session configuration is not dynamic. Before the system can recognize a new configuration, you must stop WebSphere Voice Response and start it again.

   One of the items of information that is listed for each session is the LU name. The LU name is taken from information that is provided to Communication Server as part of the network installation process. If the person who installed the network does not provide the information, the LU names are displayed as **Undefined**. Undefined LU names have no effect on session configuration.

1. From the Welcome window, select **Configuration —> 3270 Session Configuration**.

2. **Creating a session**: Click File —> New.
   
   The system displays the 3270 Session Create window.

3. Click the 3270 Link Station.
   
   The system lists all unconfigured sessions that are defined for that attachment.

4. Click one or more sessions.

5. **Assigning a server to a session**: Type in the name of the dummy server that you created to support screen capture, or click **Server Name**, click the server, and click **OK**.
   
   The system displays the 3270 Session Create window with the name of the server filled in.

6. **Saving the configuration**: Click **OK**.
   
   The system displays the 3270 Session Configuration window that lists the session you selected and shows that it is assigned to the dummy server.
7. Close the 3270 Session Configuration window.

8. Activating the session: Click Operations —> Immediate Shutdown or Quiesce Shutdown.
   The system prompts you to verify.
9. Click OK.

When you restart WebSphere Voice Response, the session is configured and ready for use by the assigned server.

Testing the 3270 connection

Use this procedure to test the 3270 configuration by starting the emulation function. To configure a session for use by the emulation function, use the procedure that is given in "Configuring a 3270 session" on page 129.

1. At the Welcome window, select Applications—> 3270 Servers.

2. Starting emulation: Click the server from the list of Servers.

3. Click Options —> Emulate.
   The system opens the Screen Capture window and displays a host screen.

4. Checking emulation: Click the Welcome window, and click Operations —> 3270 Session Manager.
   The system displays the 3270 Session Manager window. The status of the session you configured is Emulating.

5. Close the 3270 Session Manager window.

6. Close the Screen Capture window.
   The system frees the session for someone else to use.

Accommodating new 3270 servers

When you install a voice application that includes a 3270 server, you might want to change the way 3270 terminal sessions are configured. If not enough sessions are available for the new server, you might want to configure additional sessions or reconfigure any excess sessions, and assign them to the new server.

Configuring additional sessions

If the number of existing terminal sessions cannot support an additional server, review the information that is given in "How many sessions?" on page 127. This information tells you how many sessions WebSphere Voice Response can support. When you have read "How many sessions?" on page 127, follow the instructions that are given in "Configuring a 3270 session" on page 129.
Reconfiguring 3270 sessions

Use this procedure to reassign a 3270 terminal session to a different 3270 server. To configure additional sessions for use by the server, use the procedure that is given in "Configuring a 3270 session" on page 129.

Before you can reconfigure the session, ensure that the voice application developer has defined the 3270 server and that you know the name of the server.

Session reconfiguration that uses this procedure is not dynamic. You must stop WebSphere Voice Response and start it again before the voice application can use the reconfigured session. You can reconfigure a session dynamically by using the procedure that is given in "Managing 3270 Sessions" in the WebSphere Voice Response for AIX: Managing and Monitoring the System book.

1. From the Welcome window, select Configuration —> 3270 Session Configuration
2. Selecting a session: Open the session (or sessions) you want to configure for use by the new server.
   The system displays information about each session in the 3270 Session Update window. If you opened multiple sessions, the windows are stacked on top of each other.
3. Reassigning a session: Click Server Name...
   The system lists all 3270 servers.
4. Click the server that is to use this session.
5. Click OK.
   The system displays the 3270 Session Update window. The name of the new server is displayed.
6. Click OK.

The next time you start WebSphere Voice Response, the session is assigned to the new server.

Updating the configuration after changing the hostname

If you change the name of a machine that has 3270 sessions configured, you must update the WebSphere Voice Response database with the new hostname.

Note: If your network is set up as a Single System Image, you can run this command only on a server. It does not work on a client.
1. Log in as dtuser.
2. Type the following command and press Enter, to run fsupdate:
   /usr/lpp/dirTalk/tools/fsupdate
3. Type the following command and press Enter, to run `DTfix3270hostname`:
   
   DTfix3270hostname

4. When the prompt Enter host to search for is displayed, enter the old fully qualified hostname name. For example, `oldname.mycompany.com`.

5. When the prompt Enter new hostname is displayed, enter the new fully qualified hostname. For example, `newname.mycompany.com`.

   When finished, the system displays Completed and returns the command prompt.
Chapter 8. Creating and managing a single system image

You can connect together a cluster of WebSphere Voice Response systems so that they can share application and voice data. Each system then has access to all the application data that is in the cluster (such as state tables and custom servers) and all the voice data (such as voice segments and voice messages).

A cluster of WebSphere Voice Response systems that is configured like this is known as a single system image (SSI). This chapter tells you how to create and manage a single system image. It explains:

- "The components of a single system image"
- "Configuring a server node" on page 137
- "Setting up a separate voice server node" on page 139
- "Configuring a client node" on page 141
- "Verifying the configuration of a single system image" on page 143
- "Querying the configuration of a node" on page 145
- "Migrating to your single system image" on page 146
- "Changing the nodes of a single system image" on page 146
- "Monitoring the performance of a single system image" on page 150
- "More information on setting up a single system image" on page 151
- "Applying PTFs on a single system image" on page 151

This chapter also describes the commands you will use when you manage a single system image:

- "ssimkclient command" on page 152
- "ssimksvr command" on page 153
- "ssirmclient command" on page 154
- "ssirmsvr command" on page 154
- "ssistatus command" on page 155

The components of a single system image

Each system in the single system image is known as a node. You must configure each node either as a client or as a server:

Client node

A client node handles the interactions with callers. It runs WebSphere Voice Response (configured as a client), and it must have a connection
to your telephony environment. A client node contains no application
data; it gets this data from the server to which it is connected by a
local area network.

**Database server node**
A database server node contains the application object database. This
is a DB2® database that contains all the prompts and state tables that
all the WebSphere Voice Response systems in the single system image
can use. It contains also information about the custom servers that are
installed. The database server node also contains the program files for
the custom servers that are installed on the single system image. The
database server node has WebSphere Voice Response installed
(configured as a server). You can add a connection to your telephony
environment, if you want the server node to handle interactions with
some callers.

**Voice server node**
A voice server node contains the voice data for all the voice
applications that run on the single system image. The node stores its
information in an AIX file system. This node need not have
WebSphere Voice Response installed, unless you want it to handle
interactions with some callers; in this case, the node must also have a
connection to your telephony environment.

The database server and the voice server are usually on the same pSeries
computer, but you can install them onto two separate systems if you are
creating a large single system image and you want to spread the processing
load across two pSeries computers.

In comparison, a stand-alone WebSphere Voice Response system (that is, one
not configured as an SSI node) must have WebSphere Voice Response, the
telephony connection, the application data, and the voice data all installed on
the same pSeries computer. If you want to create an additional system, you
must install all these items onto a new stand-alone system.

The nodes of a single system image must be connected together using a local
area network. The type of network that you use depends on the size of the
voice solution you are implementing. For example, a small cluster running a
simple information-announcement application (such as a recording of a
weather forecast) might require only a token ring network. However, a larger
cluster that runs many voice applications or that runs a voice messaging
service, might require a network that can provide a higher capacity and
performance, such as an asynchronous transfer mode (ATM) network.

*Figure 14 on page 135* shows a stand-alone WebSphere Voice Response system.
The system is not connected to any other WebSphere Voice Response systems.
The data it uses, both application data and voice data, is stored on the same pSeries computer as WebSphere Voice Response is.

Figure 14. A Stand-alone WebSphere Voice Response system

Figure 15 shows a small single system image. Each of the clients has 6 trunks of telephony, and the server has two trunks installed. However, you do not have to install telephony components on the server. The data WebSphere Voice Response uses, both application data and voice data, is stored on the server. The single system image shown in the figure is suitable for running an IVR application.

Figure 15. A small single system image
Figure 16 shows a larger single system image. This image has more clients installed and the server has no telephony components. The data WebSphere Voice Response uses, both application data and voice data, is stored on the server. This configuration is suitable for a large voice messaging system, and it is likely that the server will perform no functions other than to serve the WebSphere Voice Response single system image.

Configuring the nodes of a single system image

When you set up a single system image, you must configure the database server node first because you need to refer to the database server when you configure a client node. You can also configure a separate server to look after your voice data, but in many installations this is not necessary. You can then configure each of the systems that will be client nodes in your single system image. Finally, you can verify that your image is working correctly. The following sections describe how to perform these tasks.
Configuring a server node

The database server node is the WebSphere Voice Response system that contains the DB2 database. All the nodes of the WebSphere Voice Response single system image use this database to store application data files, such as definitions of state tables and voice prompts. It also contains source, binary, and data files for custom servers.

In most installations you will want to keep the voice data on this system also, so this section describes how to configure a server node that contains both the WebSphere Voice Response database and the voice server. If you want to set up the voice server on a separate node (for example, if you are setting up a large voice messaging system), see "Setting up a separate voice server node" on page 139.

To access the voice server file system, applications must use Network File System (NFS); to access the DB2 database, applications must use the DB2 application programming interface.

This section describes the basic NFS commands that you need to use when configuring your single system image. For more information on NFS, see the System Management Guide; Communications and Networks book.

Before you start

Before you start configuring the server node, ensure that the following are true:

* The AIX account that WebSphere Voice Response will use (usually this is named dtuser) must be set up identically on every node of the single system image. For information on how to do this, refer to the WebSphere Voice Response for AIX: Installation book.
* WebSphere Voice Response is already installed on the node.
* The WebSphere Voice Response database is already created on the node (for information on how to do this, refer to the WebSphere Voice Response for AIX: Installation book).
* WebSphere Voice Response is not running on the node. If it is, use the DT_shutdown command to stop it.

Creating the server

On the system you want to configure as a server:

1. Log in as the WebSphere Voice Response account (usually this is dtuser).
2. Respond with 2 when prompted, to prevent WebSphere Voice Response from starting.
3. Run the ssimksvr command.
   For more information on this command, see "ssimksvr command" on page 153.
Identifying the client nodes

After you have configured the server, you must export the custom server information and voice files so they can be mounted on the client nodes. To do this:

1. Log in as root by typing the command su root
2. Export the custom server information by typing the following NFS command:
   
   mknfsexp -d /home/dirTalk/current_dir/ca.local
   -c client1,client2,...clientN
   
   where client1 through clientN is a comma-separated list of the host names of the client nodes of your single system image.
3. Export the voice files by typing the following NFS command:
   
   mknfsexp -d /home/dirTalk/current_dir/voice.local
   -c client1,client2,...clientN
   
   where client1 through clientN is a comma-separated list of the host names of the client nodes of your single system image.

The mknfsexp command saves the export definition in the /etc/exports file, so that the directories are exported every time the database server starts.

Note: The mknfsexp command shown here is in its simplest form. If you already use NFS on your installation, you might want to specify other flags on this command so that you maintain the integrity of your system. For more information on this command, see the Commands Reference.

Checklist for configuring a server

To ensure that you have performed all the steps required to configure a server node that contains both the WebSphere Voice Response database and the voice server, use this checklist:

- Log in as dtuser.
- ssimksvr command.
- Run the Log in as root.
- Export the custom server information for mounting on the clients (using the mknfsexp command).
- Export the voice files for mounting on the clients (using the mknfsexp command).
Setting up a separate voice server node

You might want to store your voice data (for example, voice segments, user greetings, and voice messages) on a server that is reserved for that purpose. This might be desired if you have a large single system image that is running a voice messaging service and you want to reserve a separate server to store your voice data. You do not need to install WebSphere Voice Response on this server.

**Note:** If you want to store your voice data on the database server, you can skip this section; you do not need to perform any additional tasks to set up your server.

Voice data is stored in data files in an AIX file system. In a single system image, this file system is mounted onto the client nodes using NFS. To set up a separate voice server node, you must create the file system to store the voice data, and export that file system so that the client nodes can access it. You must also set up the database server, as described in "Configuring a server node" on page 137.

**Note:** If you already use NFS on your installation, you might want to modify the instructions given here so that you maintain the integrity of your system. For more information, see the System Management Guide; Communications and Networks book.

**Configuring the voice server**

On the system that you want to configure as a voice server:

1. Log in as root.
2. Create and mount a new file system to store your voice data.
   - The size of this file system is determined by the amount of voice data (including voice messages, voice segments, and user greetings) that you expect your single system image to generate.
   - For information on how to create and mount a file system, see the System Management Guide; Communications and Networks book.
   - The following instructions assume that you name your new file system /ssi.
3. Copy the file /home/dirtalk/DIRTALK.vox.tar.Z (which is supplied with WebSphere Voice Response) to the server you are working on.
   - The following instructions assume that you copy this file to the /tmp directory.
4. Type the following command:
   - cd /ssi
5. Type the following command:
   - zcat /tmp/DIRTALK.vox.tar.Z | tar -xvf-
6. Type the following command:
   ```bash
   chown -R dtuser:staff /ssi
   ```

7. Type the following NFS command:
   ```bash
   mknfsexp -d /ssi -c client1, client2,...clientN, database_server -r database_server
   ```
   where client1 through clientN is a comma-separated list of the host names of the client nodes of your single system image, and database_server is the host name of the database server.
   This command saves the export definition in the /etc/exports file, so the directory is exported every time the server starts.

   **Note:** The mknfsexp command shown here has the -r flag specified to allow the database server to access files on this partition as a root user. The -r flag is required for saveDT to function correctly. For more information about this command, see the Commands Reference.

**Changing the database server to work with the voice server**

When you have configured the voice server node, you must go back to the database server to make it work with your voice server.

On the database server node, perform the following steps:

1. Log in as the WebSphere Voice Response account (usually this is dtuser).
2. Respond with 2 when prompted, to prevent WebSphere Voice Response from starting.
3. Log in as root by typing the command `su root`
4. Type the following NFS command to set up the NFS mounts to the WebSphere Voice Response database:
   ```bash
   mknfsmnt -f /home/dirTalk/current_dir/voice.ssi
            -d voice_directory -h voice_server -m ssi -S -p 32 -A
   ```
   where voice_directory is the directory you created when you set up the voice server (for example, /ssi) and voice_server is the host name of the voice server.

   **Note:** The mknfsmnt command shown here is in its simplest form. If you already use NFS on your installation, you might want to specify other flags on this command so that you maintain the integrity of your system. For more information on this command, see the Commands Reference.

5. Type `exit` to return to the dtuser account.
6. Type the following command:
   ```bash
   /usr/lpp/dirTalk/tools/fsupdate
   ```
Configuring a client node

A client node is a WebSphere Voice Response system that can operate only when it is connected to a database server as part of a single system image. Typically you will want to have many clients in your single system image; you must perform the tasks described in this section on each one.

Before you start

Before you start configuring a client node, ensure that the following are true:

- The AIX account that WebSphere Voice Response will use (usually this is named dtuser) is set up identically on every node of the single system image. For information on how to do this, see the WebSphere Voice Response for AIX: Installation book.
- WebSphere Voice Response is already installed on the node.
- WebSphere Voice Response is not running on the node.
- The database server node (and the voice server node, if you want this to be separate) is already configured.

Configuring the client

On the system you want to configure as a client:

1. Log in as the WebSphere Voice Response account (usually this is dtuser).
2. Respond with 2 when prompted, to prevent WebSphere Voice Response from starting.
3. Type the following command:
   
   ssimkclient server_hostname

   where server_hostname is the host name of the database server node.

   You are prompted to type the password of the WebSphere Voice Response user on the server. This ensures that a connection can be made to that server. Normally this login ID is dtuser.

   For more information, see “ssimkclient command” on page 152.

Identifying the servers

After you have configured the client, you must set up the NFS mounts to provide access to shared custom server information. You must also set up access to the voice server, if that is separate. To do this:

1. Log in as root by typing the command su root
2. Type the following NFS command to create the NFS mount for the custom server directories:

   mknfsmnt
   -f /home/dirTalk/current_dir/ca.ssi
   -d /home/dirTalk/current_dir/ca.local
   -h server
   -m ssi
   -H
On the -h flag, for server specify the host name of the database server.

Use the -H option above to hard mount the custom server directories. This is important because allows WebSphere Voice Response to mount the file systems when required. If the /ca directory is inaccessible because it is set to soft, the AIX Virtual Memory Manager causes a bus error that causes the custom server to core dump. When the mount is set to hard, the custom server waits until the /ca directory becomes available.

For more information, see "The mknfsmnt command."

3. Type the following NFS command to create the NFS mount for the voice files:

```
mknfsmnt
-f /home/dirTalk/current_dir/voice.ssi
-d /home/dirTalk/current_dir/voice.local
-h server
-m ssi
-S
-p 32
-A
-w bg
-o 10
-K 3
-k udp
-R 2
```

On the -h flag, for server specify the host name of the network interface that you want to use for NFS data on the voice server.

Use the -S option above to soft mount the voice files so that requests to play voice segments can time-out if the server is unavailable.

Note: If you are configuring your database and voice servers on the same pSeries computer, specify the host name of its network interface on the -h flag.

For more information, see "The mknfsmnt command."

**The mknfsmnt command**

If you already use NFS on your installation, you might want to specify other flags on the mknfsmnt command so that you maintain the integrity of your system. In the commands shown above, the flags are used like this:

- **-m** Identifies a mount type for Single System Image
For more information on this NFS command, see the *Commands Reference*.

**Checklist for configuring a client**

To ensure that you have performed all the steps required to configure a client node, use this checklist:

- Log in as `dtuser`.
- Run the `ssimkclient` command, specifying the host name of the database server.
- Log in as `root`.
- Create a mount for the custom server directories (using the `mknfsmnt` command).
- Create a mount for the voice files (using the `mknfsmnt` command).

**Verifying the configuration of a single system image**

After you have configured the nodes of your single system image, verify that it is set up correctly. Your first step is to run the `ssistatus` command on each node of your single system image. For more information on this, see "Querying the configuration of a node" on page 145.

Follow the instructions in this section to display a voice segment on the server, copy a voice segment into a server directory, then display that segment on a client.

**Verifying the server node**

Perform the following steps on the server:

1. Start WebSphere Voice Response.
   - If the WebSphere Voice Response Status window shows any problems, look in the error log for more information.
2. Start the System Monitor:
   - `Welcome window` --> `Operations` --> `System Monitor`
3. Click one of the supplied application profiles:
   - `Welcome window` --> `Configuration` --> `Application Profiles`
4. Click one of the profiles that are listed, then:
5. Open the Voice Segment Editor:
   Welcome window —> Applications —> Voice Segments
6. Click the U.S. English language.
7. Click the System voice directory.
8. Click segment 1 One, then Segment —> Open
   If the voice segment is displayed, the database connection is set up correctly and applications can communicate with the voice server.
   If the voice segment is not displayed, an alarm should be generated, indicating that the NFS export and mounts have not been set up correctly.

1. If any of the steps above fail: Stop WebSphere Voice Response by using the DT_shutdown command.
2. Retry the verification procedure.

Prepare a voice segment
Before you verify each of your client nodes, prepare a voice segment that you can try to access from each client.

On your voice server node:
1. Create a new voice directory and copy an existing voice segment into it.
2. In the Voice Directories and Segments window, click Directory —> New
3. Create a new voice directory by typing any name in the dialog.
4. In the Voice Segment window, click File —> Save As
5. Click the voice directory that you created in Step 1, then type an ID for the segment (for example, 1).
   When you click OK, the voice segment is copied into the new voice directory.

Verifying a client node
On each client node in your single system image:
1. Start WebSphere Voice Response.
   If the WebSphere Voice Response Status window shows any problems, look in the error log for more information.
2. Start the System Monitor:
   Welcome window —> Operations —> System Monitor
3. Click one of the supplied application profiles:
   Welcome window —> Configuration —> Application Profiles
4. Click one of the profiles that are listed, then:
   File —> Open
5. Open the Voice Segment Editor:
Welcome window —> Applications —> Voice Segments

6. Click the U.S. English language.
7. Click the System voice directory.
8. Click segment **1 One** then, Segment —> Open
   
   If the voice segment is displayed, this means that the database connection is set up correctly and that applications can communicate with the voice server.
   
   If the voice segment is not displayed, an alarm should be generated, indicating that the NFS export and mounts have not been set up correctly.

9. Click the segment that you created in “Prepare a voice segment” on page 144, then Segment —> Open
   
   If the voice segment is displayed, this means that the client can communicate with the voice server.
   
   If the voice segment is not displayed, an alarm should be generated, indicating that the NFS export and mounts have not been set up correctly.

If any of the steps above fail:
1. Stop WebSphere Voice Response on all nodes of the single system image by using the DT_shutdown command.
2. Retry the verification procedure.

---

**Querying the configuration of a node**

To find out if a WebSphere Voice Response system is configured as a client node, as a database server node, or as a stand-alone system, use the ssistatus command.

The ssistatus command is described in “ssistatus command” on page 155.

When you run the ssistatus command on a server node, you get results like this:

```
ssistatus: Querying configuration..
ssistatus: cleese.mydomain.com is a database server for clients:
HOSTNAME
----------------------------------------
savana.mydomain.com
idle.mydomain.com
tarbuck.mydomain.com
sp5tr1.mydomain.com
sp5tr2.mydomain.com
5 record(s) selected.
```

When you run the ssistatus command on a client node, you get results like this:
When you run the `ssistatus` command on a stand-alone WebSphere Voice Response system, you get results like this:

```
ssistatus: Querying configuration ..
ssistatus: rum.mydomain.com is a Standalone system
```

---

**Migrating to your single system image**

If you have an existing WebSphere Voice Response system from which you have saved your data, you are now ready to migrate that data to your new single system image. To do this, use the save-and-restore method of migration, which is described in the WebSphere Voice Response *WebSphere Voice Response for AIX: Installation* book.

---

**Changing the nodes of a single system image**

After you have configured your single system image and started using it, you might need to change some of the nodes in the cluster. The following sections describe to:

- Remove a client from the single system image
- Remove a server from the single system image
- Add a new client to your single system image
- Change the password on the server
- Change the network properties of a node

**Removing a client from the single system image**

You might want to remove a client node from the single system image temporarily while you perform some maintenance on the system hardware. To do this, you must first ensure that:

- The system that contains the client node is running
- WebSphere Voice Response is not running

Then, on the client node:

1. Log in as the WebSphere Voice Response account (usually this is `dtuser`).
2. Respond with 2 when prompted not to start WebSphere Voice Response.
3. Run the `ssirmclient` command.
   If you want more information, see “ssirmclient command” on page 154.

The system is now a stand-alone WebSphere Voice Response system.

When you want to use the system as a client node again, type the following command:

```plaintext
ssimkclient server_hostname
```

where `server_hostname` is the host name of the database server node. You are prompted to type the password of the WebSphere Voice Response user on the server. This ensures that a connection can be made to that server. Normally this login ID is `dtuser`. For more information, see “ssimkclient command” on page 152.

If you want to use the system permanently as a stand-alone WebSphere Voice Response system:

1. Perform the steps to remove the client from the single system image.
2. On the system that was configured as a client, log in as `root` by typing the command `su root`
3. Remove the NFS mounts from the client node by typing the following NFS commands:
   ```plaintext
   rmnfsmnt -f /home/dirTalk/current_dir/ca.ssi
   rmnfsmnt -f /home/dirTalk/current_dir/voice.ssi
   ```
   For more information on this NFS command, see the Commands Reference.
4. On the SSI server node, remove the host name of the client node from the access list that is on the exports that you created when you configured the client node. To do this:
   a. Log in as `root`.
   b. Log in as `/etc/exports`
   c. Find the lines that WebSphere Voice Response uses.
      For example:
      ```plaintext
      /home/dirTalk/current_dir/ca.local-access=starbuck.mydomain.com
      ```
   d. Delete the host name of the client that you are removing from the single system image.

   **Note:** If you are removing the only client from the single system image, you must delete the whole line.
   e. Type the following NFS commands to stop the file systems from being available to the client:
      ```plaintext
      exportfs -u /home/dirTalk/current_dir/ca.local
      exportfs -u /home/dirTalk/current_dir/voice.local
      ```
Removing a server from the single system image

If you want to remove a server node from the single system image, perhaps only while you perform some maintenance on its hardware, you must unconfigure it from the single system image.

1. Remove all the client nodes that use the server (see "Removing a client from the single system image" on page 146).
2. Ensure that:
   - The system that contains the server node is running
   - WebSphere Voice Response is not running
3. On the server node:
   a. Log in as the WebSphere Voice Response account (usually this is dtuser).
   b. Respond with 2 when prompted, to prevent WebSphere Voice Response from starting.
   c. Run the ssirmsvr command.
      If you want more information, see "ssirmsvr command" on page 154.

The system is now a stand-alone WebSphere Voice Response system. When you want to use the system as a server node again, follow the instructions that are given in "Configuring a server node" on page 137.

If you never want to use the system as a server node again (that is, you want to use it permanently as a stand-alone WebSphere Voice Response system), you must also remove the NFS exports. To do this, type the following NFS commands:

rmnfsexp -d /home/dirTalk/current_dir/ca.local
rmnfsexp -d /home/dirTalk/current_dir/voice.local

Adding a new client to your single system image

After you have set up your single system image and started using it, you might need to add another client. You must identify this new client to the voice server and export the voice database directory to the new client. To do this:

1. Log in as root on the voice server.
   The voice server is on the pSeries computer that contains your database server, unless you have set up a separate voice server.
2. Add the host name of the client node to the access list that is on the exports that you created when you configured the client node. To do this:
   a. Edit the file /etc/exports
   b. Find the lines that WebSphere Voice Response uses.
      For example:
      /home/dirTalk/current_dir/ca.local -access=taruck.mydomain.com
c. Add the host name of the client that you are adding to the single system image.

For example:

/home/dirTalk/current_dir/ca.local -access=tarbuck.mydomain.com:idle.mydomain.com

Note: In your file, ensure that this remains as a single line.

d. Type the following NFS command to make the file system available to the new client:

```
exportfs -av
```

3. Configure the client node as described in “Configuring a client node” on page 141.

What happens if you change the password on a server

When you configure a client node in a single system image, you must specify the AIX login user ID and password of the WebSphere Voice Response account on the database server node. The user ID is normally dtuser.

To change the password on the server:

1. Unconfigure each client node (see “Removing a client from the single system image” on page 146).
2. Change the password on the server.
3. Reconfigure each client node, specifying the new password (see “Configuring a client node” on page 141).

What happens if you change the network configuration of a node?

When you configure a client node in a single system image, you must specify the host name of the database server node.

This means that if you have to change the host name of the server, you must unconfigure each client node (see “Removing a client from the single system image” on page 146), then reconfigure each client node, specifying the new name or address (see “Configuring a client node” on page 141).

Changing the number of database connections

The maximum number of concurrent DB2 applications is determined by the DB2 maxappl parameter. The default value is 40. If the value of this parameter is too low, WebSphere Voice Response processes might not be able to connect to the WebSphere Voice Response database.

You might need to change the value of this parameter on the database server node of a single system image:

- When you add new client nodes to the single system image.
- When you increase the value of the Number of Voice Messaging Servers system parameter.
When you add new client nodes, increase the value of the `maxappls` parameter by 20 for each new client.

To increase the value of the `maxappls` parameter:
1. Login to the database server node as `root`.
2. Change to the DB2 instance ID for WebSphere Voice Response by typing the command:
   ```bash
   su - dtdb23in
   ```
3. Change the value of the parameter by typing the command:
   ```bash
   db2 update database configuration for dtdbv230 using maxappls n
   where n is the new value.
   ```

---

Monitoring the performance of a single system image

On a single system image, the voice data for all the applications that are running on the image is stored on a single voice server. A caller’s telephone is connected to a client node that is on the single system image, so WebSphere Voice Response must retrieve voice data from the server, and send it to the client, before the application can play it to the caller. This means that the efficiency of the network that connects the client to the server has a large effect on the performance of the application.

You can improve the performance by using a high-performance network that serves only your single system image (that is, the network does not carry data for applications other than those that serve the WebSphere Voice Response system). It also helps if you design applications that use state table actions such as `PlayPrompt`, which play voice data that is cached by WebSphere Voice Response. In contrast, the `PlayVoiceSegment` action always retrieves voice data from the database.

When your applications are running, you will want to monitor their performance to ensure that your callers do not receive poor quality. WebSphere Voice Response helps you by measuring the following attributes, which affect how voice data is played:
- Play Latency Time (PLT)
- Underrun Margin Time (UMT)
- Check Voice Messages Time (CHK)
- Profile Retrieval Time (PRF)

For each of these attributes, WebSphere Voice Response records a measurement every time it performs an action that affects the attribute. You can configure WebSphere Voice Response so that it issues warning messages when the values of these measurements reach thresholds that you specify. You can also use the `DTmon` command to display the most recent measurement.
for each attribute. For more information on how to do this, see the *WebSphere Voice Response for AIX: Managing and Monitoring the System* book.

---

**More information on setting up a single system image**

For information on tuning and configuring a single system image for better performance, ask your IBM representative.

---

**Applying PTFs on a single system image**

When you want to apply a program temporary fix (PTF) to a WebSphere Voice Response system that is part of a single system image, you must first read the `.info` file that is supplied with the PTF to see if you need to apply that PTF on all the nodes of the single system image.

**Applying the PTF on some nodes**

If the `.info` file that is supplied with the PTF says you do not have to apply the PTF on all the nodes of the single system image, you can apply it on only those nodes you choose. This means that the fixes delivered in the PTF are available only on those nodes; the WebSphere Voice Response code on the other nodes of the single system image remains at the previous level.

The procedure you must follow to apply the PTF on a node depends on whether that node is a client or a server:

- **To apply the PTF on a client node:**
  1. Shut down WebSphere Voice Response on that node.
  2. Apply the PTF on that node.
  3. Shutdown and restart the pSeries computer on that node.
  4. Restart WebSphere Voice Response on that node.

- **To apply the PTF on the server node:**
  1. Shut down WebSphere Voice Response on all the client nodes of the single system image.
  2. Shut down WebSphere Voice Response on the server node.
  3. Apply the PTF on the server node.
  4. Shutdown and restart the pSeries computer on the server node.
  5. Start WebSphere Voice Response on the server.

**Applying the PTF on all nodes**

If the `.info` file that is supplied with the PTF says you must apply the PTF on all the nodes of the single system image, the fixes delivered in the PTF become available on all nodes, and the WebSphere Voice Response code is at the same level on all nodes.
To apply the PTF:
1. Shut down WebSphere Voice Response on all the client nodes.
2. Shut down WebSphere Voice Response on the server node.
3. Apply the PTF on every node.
4. Shutdown and restart the pSeries computer at every node.
5. Start WebSphere Voice Response on the server.

Commands

**ssimkclient command**

**Purpose**

Configures a WebSphere Voice Response system as a client node in a single system image.

**Syntax**

```
ssimkclient server_hostname [server_port]
```

**Description**

When you run the `ssimkclient` command on a WebSphere Voice Response system, it configures that system as a client node. Before you use this command:

- The system on which you want to run the `ssimkclient` command must not be configured as a server node; if it is, the command fails.
- The server to which you want the client to connect must already be configured as a server node. To do this, use the `ssimksvr` command (see “ssimksvr command” on page 153).

When you run the `ssimkclient` command, it prompts you to type the password of the WebSphere Voice Response user on the server. This ensures that a connection can be made to that server. Normally this login ID is dtuser.

**Flags**

*server_hostname*

The host name of the network interface that you want to use for DB2 data on the WebSphere Voice Response system that is configured as the database server node in the single system image.

You must specify a fully-qualified domain name (for example, john.mydomain.com).
server_port

The TCP/IP port address (on the server) that you want to use to carry DB2 requests from the client to the server.

If you do not specify an address, WebSphere Voice Response uses the default value 50110.

If you do not want to specify a particular address, let WebSphere Voice Response use its default address. The only requirement is that no other software running on the system can use the same address.

Exit status

0 Successful completion.

>0 An error occurred. Messages show the reason for the error.

Examples

The following example configures as a client node the system on which the command is run. The client connects to the server named john.mydomain.com using port 50110:

```bash
ssimkclient john.mydomain.com 50110
```

ssimksvr command

Purpose

Configures a WebSphere Voice Response system as a server node in a single system image.

Syntax

```bash
ssimksvr [server_port]
```

Description

The `ssimksvr` command on a WebSphere Voice Response system, it configures that system as a server node. Before you use this command, the system on which you want to run the `ssimksvr` command must not be configured as a client node; if it is, the command fails.

Flags

`server_port`

The TCP/IP port address (on the server) that you want to use to carry DB2 requests from the clients to the server.

If you do not specify an address, WebSphere Voice Response uses the default value 50110.
If you do not want to specify a particular address, let WebSphere Voice Response use its default address. The only requirement is that no other software running on the system can use the same address.

**Exit status**

0 Successful completion.

>0 An error occurred. Messages show the reason for the error.

**ssirmclient command**

**Purpose**

Removes a client node from a single system image.

**Syntax**

ssirmclient [-f]

**Description**

When you run the `ssirmclient` command on a WebSphere Voice Response system that is configured as a client node in a single system image, it removes that system from the single system image. On successful completion of this command, the system is a stand-alone system; that is, it is no longer configured to communicate with any other WebSphere Voice Response systems.

**Flags**

- `-f` Remove the client node from the single system image even if an error occurs.

**Exit status**

0 Successful completion.

>0 An error occurred. Messages show the reason for the error.

**ssirmsvr command**

**Purpose**

Removes a server node from a single system image.

**Syntax**

ssirmsvr [-f]
Description

When you run the ssirmsvr command on a WebSphere Voice Response system that is configured as a server node in a single system image, it removes that system from the single system image. On successful completion of this command, the system is a stand-alone system; that is, it is no longer configured to communicate with any other WebSphere Voice Response systems.

Flags

- `f` Remove the server node from the single system image even if an error occurs.

Exit status

- `0` Successful completion.
- `>0` An error occurred. Messages show the reason for the error.

ssistatus command

Purpose

Displays how the WebSphere Voice Response system is configured within the single system image.

Syntax

 `ssistatus` 

Description

When you run the ssistatus command on a WebSphere Voice Response system, it displays how that system is configured within a single system image. The information that the command shows depends on how the system is configured:

For a client node

The command shows the database server catalog entry.

For a database server node

The command shows information about the client nodes that are configured to communicate with the server.

For a stand-alone system

The command shows that the system is not configured as part of a single system image.
This command also tests that the system on which the command is run has access to the AIX files that store the voice data.

You cannot use this command on a WebSphere Voice Response system that is configured only as a voice server.

For examples of the output of this command, see “Querying the configuration of a node” on page 145.

**Exit status**

<table>
<thead>
<tr>
<th>Status</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Successful completion; the system is stand-alone (that is, it is not part of a single system image).</td>
</tr>
<tr>
<td>1</td>
<td>Successful completion; the system is a client node.</td>
</tr>
<tr>
<td>2</td>
<td>Successful completion; the system is a server node.</td>
</tr>
<tr>
<td>255</td>
<td>An error occurred. Messages show the reason for the error.</td>
</tr>
</tbody>
</table>
Chapter 9. Adding languages

This chapter describes how to define additional languages to be used either by voice applications or for translated window text.

About additional languages

WebSphere Voice Response can be set up to operate in more than one language. One WebSphere Voice Response voice application can play voice to callers in multiple languages. The WebSphere Voice Response window text can display in different languages to different people at the same time. In addition, the keyboard for the pSeries computer can use the character sets for different languages.

To find out which languages are delivered with this release of WebSphere Voice Response, see the README file in /usr/lpp/dirTalk/readme.

Why do I need more languages?

When you first install WebSphere Voice Response, the only language available is U.S. English. Defining additional languages enables WebSphere Voice Response to operate in other languages.

Voice segments

When you define additional languages, WebSphere Voice Response can store voice segments in different languages in separate voice databases. When voice segments are stored in language-specific voice databases, they can be found more easily. (For an introduction to creating and storing voice segments, see the WebSphere Voice Response for AIX: Designing and Managing State Table Applications book.)

Unless you define additional languages, all voice segments are stored in one database: the database for U.S. English.

System prompts

When you define additional languages, voice application developers can translate the system prompts, or use the non-U.S. English system prompts that are delivered with WebSphere Voice Response. The number that the system prompts receive as input is played differently by each particular system prompt. For example, one prompt plays the number as the date. Another plays the number as an amount of currency. The syntax and semantics for
saying numbers, dates, and times is different for each language, so the system prompts must be changed if you decide to use multiple languages.

The system prompts are described in detail in the *WebSphere Voice Response for AIX: Application Development using State Tables* book. For latest information about the non-U.S. English prompts supplied with the product, see the README file in /usr/lpp/dirTalk/readme.

**Window text**

When you define additional languages, WebSphere Voice Response can maintain multiple copies of the text that is displayed in the windows, including the online help text. Each copy of the text is stored in a language-specific text database and can be translated into that language.

The language in which WebSphere Voice Response displays window text when someone logs on is determined by the language that is specified in the user's administrator profile. Unless you define additional languages, you can never translate the window text. Everyone will have to use the system in English.

**How do I get more languages?**

For the system to operate in another language, you must do the following:

- Always add the language (described in "Defining additional languages" on page 162).
- If you need window text in the language, translate window text into the new language (described in "Using WebSphere Voice Response to translate window text" on page 166 and "Using another editor to translate display text" on page 172).
- If people want the translated window text to display in the windows, add administrator profiles that specify the language as the preferred language (described in "Giving people access to WebSphere Voice Response" on page 5).
- If the language is supplied, import it. If the language is not supplied, record new voice segments and create new system prompts as required. See the *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* book for information about importing languages and for an introduction to voice segments and prompts.
- If the system prompts are not available in the new language but voice application developers need them, translate the prompts (described in the *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* book).
What defines a new language?

A new language is defined by associating a language name with a locale and identifying the type of language.

The language name is what you use to identify the language. The locale identifies a language-territory combination and is used to define particular system conventions, such as the format of date-time stamps.

The language type describes how you intend to use the language. Three types of language are available: Voice, Window Text, and Window Text and Voice. If you want to record voice segments in the language, the language is a Voice language. If you want to translate window text into the language, the language is a Window Text language. If you want to do both, the language is a Window Text and Voice language.

When you define a new language, WebSphere Voice Response assigns it a code. The code is a number between 1 and 255 that the system uses to identify the language. When the system displays a list of languages, it displays the number to the left of the name of the language.

How many languages can I define?

WebSphere Voice Response holds up to 254 language definitions. The system already includes 20 language names:

- Danish
- Belgian Dutch
- Netherlands Dutch
- U.K. English
- U.S. English
- Finnish
- Belgian French
- Canadian French
- French
- Swiss French
- German
- Swiss German
- Greek
- Icelandic
- Italian
- Norwegian
- Portuguese
- Spanish
- Swedish
- Turkish.

Each language name can identify only one language at a time. When you have labeled a language with one of the names, the name is no longer available for a different language. You can also create your own names for languages.

The system also includes 20 locales:

<table>
<thead>
<tr>
<th>Locale</th>
<th>Language/Territory</th>
<th>Locale</th>
<th>Language/Territory</th>
</tr>
</thead>
<tbody>
<tr>
<td>da_DK</td>
<td>Danish/Denmark</td>
<td>de_DE</td>
<td>German/Germany</td>
</tr>
<tr>
<td>nl_BE</td>
<td>Dutch/Belgium</td>
<td>de_CH</td>
<td>German/Switzerland</td>
</tr>
<tr>
<td>nl_NL</td>
<td>Dutch/Netherlands</td>
<td>el_GR</td>
<td>Greek/Greece</td>
</tr>
<tr>
<td>en_GB</td>
<td>English/Great Britain</td>
<td>is_IS</td>
<td>Icelandic/Iceland</td>
</tr>
<tr>
<td>en_US</td>
<td>English/United States</td>
<td>it_IT</td>
<td>Italian/Italy</td>
</tr>
<tr>
<td>fi_FI</td>
<td>Finnish/Finland</td>
<td>no_NO</td>
<td>Norwegian/Norway</td>
</tr>
<tr>
<td>fr_BE</td>
<td>French/Belgium</td>
<td>pt_PT</td>
<td>Portuguese/Portugal</td>
</tr>
<tr>
<td>fr_CA</td>
<td>French/Canada</td>
<td>es_ES</td>
<td>Spanish/Spain</td>
</tr>
<tr>
<td>fr_FR</td>
<td>French/France</td>
<td>sv_SE</td>
<td>Swedish/Sweden</td>
</tr>
<tr>
<td>Fr_CH</td>
<td>French/Switzerland</td>
<td>tr_TR</td>
<td>Turkish/Turkey</td>
</tr>
</tbody>
</table>

Any locale can be associated with any language name to define the language. For example, the system does not stop you from defining a language called Icelandic with a locale called da_DK.
About TDD languages

An additional language name **U.S. English TDD** is defined to support applications that interact with Telecommunications Devices for the Deaf. If you have installed the optional Telecommunications Devices for the Deaf package, you must import the supplied voice segments and system prompts. These segments and prompts provide TDD equivalents for the base system prompts and segments.

To add the TDD languages, follow the instructions that are given in "Defining additional languages" on page 162. Choose a Language Type of **Voice Only**, and a New Language Name of **U.S. English TDD**. Do not copy any voice information from other languages, because this information is not needed and uses space in the voice database.

When you have saved the new language, import it (as described in the *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* book) to add the base system prompts and segments for the TDD language. The import file is `/usr/lpp/dirTalk/sw/samples/TDD.imp`.

You can now run the predefined applications for a TDD caller, simply by creating and using an application profile that specifies the **U.S. English TDD** language. *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* book describes how you can create versions of your own applications for TDD callers.

Introducing the language database

When you define a new language, WebSphere Voice Response copies an existing database to create a database for information in the language.

**What database is copied?**

You decide which language database WebSphere Voice Response copies to create the new language database. When the system is installed, the only database available is the database for U.S. English. When you have defined more languages, however, you can copy any one of them. You do not have to copy English.

For example, suppose you need two new languages: German and Swiss German. To create the German database, you must copy the U.S. English database. To create the Swiss German database, you can copy either the U.S. English database or the German database.

**What does the system copy?**

The language type that you assign to the language determines what the system copies. If you define the language as a Window Text language, the system copies the portion of the database containing the text that is displayed.
on the windows. If you define the language as a Voice language, the system copies your choice of voice tables and voice segment directories. (If you want, you can also create an empty database for a Voice language.) If you define the language as a Voice and Window Text language, the system copies both. You can copy the voice portion of the database from one language and the window text portion from another.

When you define a language as Window Text, the window text in the "new" language has to be translated before it is really in the new language. Until you translate the text, it remains an exact copy of the existing text in the "old" language, labelled as a different language.

**Defining additional languages**

Use this procedure to define an additional language.

You can copy the whole voice database, part of it, or none of it. The system lets you select individual voice directories, voice tables, or both, to copy.

To define a new language, you can associate any language name with any locale. For example, you can define a new language by associating the name U.S. English with the locale Portuguese in Portugal (pt_PT).

When you create your own name for a language, the system automatically assigns the language the highest numbered code available. For example, the first language to which you assign your own name receives the code 255.

To change the code to an unused code, use the following procedure.

**Procedure**

1. From the Welcome window, select Configuration —> Languages.
2. **Listing the existing languages:** To see all the existing languages, ensure that the displayed list of languages is titled "All Languages." If it is not, click View —> All Languages.
   The system lists all the languages in the system.

3. To see only the languages of one type, click View —> Window Text Languages or Voice Languages.
   The system lists all languages of the type you selected.
   The system displays the Language window:

   ![Language window](image)

5. Defining the language type: Click the button that is next to the language type that applies to the new language.

6. Click File —> Save.
   The system displays the Language Save As window:

   ![Language Save As window](image)

7. Selecting the voice database to copy: If the language is a Voice language, click Source for Voice....
The system lists all languages that are now defined as either Voice Only, or Voice and Window Text languages.

8. Click the language database to copy.
9. Click OK.
   The system displays the Language Save As window. If the new language is also a Window Text language, the source language that is chosen for voice is filled in as the Source for Window Text.
10. Selecting directories: To select individual voice directories to copy, click the button that is next to Voice Directories.
    The system lists all directories in the selected database.

11. Click the directories that are to be copied.
12. Copying everything: If you want to copy all the directories and tables, click Select All.
13. Selecting a window text database to copy: To change the window text database, click Source for Window Text....
14. The system lists all languages that are now defined as either Window Text or Voice and Window Text languages.
15. Click the language database that you want to copy.
16. Click OK.
17. Naming the new language: Click New Language Name...
    The system lists the available language names.
18. **Using an existing name:** To use an existing name for the language, click the name you want to assign to the new language.

19. Click **OK**.
    The system assigns the name to the new language.

20. **Creating your own name:** To create your own name for the language, type the name in the **User Defined** field.

21. Click **OK**.
    The system displays the Language Save As window showing the source languages, the new language name, and the language code.

22. **Changing the code:** To change the code for a language you named, click the code, delete it, and type a new code.

23. **Saving the definition:** Click **OK**.
    If you named the language yourself and changed the language code, but another language is already identified by the same code, the system displays the available codes. Otherwise, it assigns the new code to the language. Without reference to how you named the language, the system adds the language. This operation might take a few minutes.

24. **Changing the locale:** To change the locale, click **Locale**...
    The system displays a list of locales.

25. Click the locale.

26. Click **OK**.
    The system displays the Language window with the new locale.

27. **Save** the new language.
Although the Language window remains displayed, the system has saved the new language definition and created the requested database. When you **Close** the Language window, you see the new language on the list of languages.

### Introducing window text

Window text includes **display** text and **help** text. Display text is all the text that is **displayed** in the window, such as field labels and button labels. Help text is the information you can access by clicking **Help** from each window, or by using the help index.

Window text does not include any text that you enter, such as the descriptions of voice segments or the labels for states in a state table.

When WebSphere Voice Response is installed, all the window text is in English. The text, however, can be translated into another language if required.

You can translate window text into any additional language for which a window text database exists. For more information on additional languages, see “[About additional languages](#)” on page 157.

### How do I translate the window text?

You can translate the display text and help text by using the text editors that are provided with WebSphere Voice Response. “[Using WebSphere Voice Response to translate display text](#)” on page 168 and “[Using WebSphere Voice Response to translate help text](#)” on page 171 describe how to use the WebSphere Voice Response editors. If you want to translate the display text, but not the help text, you can copy the display text to the hard disk on the pSeries computer and use another text editor to translate. “[Using another editor to translate display text](#)” on page 172 describes how to **export** display text for translation and **import** it when it is translated.

### Displaying window text in another language

The language in which window text is displayed after a person logs on to WebSphere Voice Response is determined by information in that person’s administrator profile. “[Giving people access to WebSphere Voice Response](#)” on page 5 describes how to create an administrator profile that allows each person to see windows that display text in the required language.

### Using WebSphere Voice Response to translate window text

WebSphere Voice Response includes two editors that you can use to translate the window text. The text string editor changes display text. The help text editor changes help text.
**Introducing display text**

Display text is organized into groups of terms. A term can be a single word (such as a button label) or a phrase (such as an instruction). Each group of terms is identified by the WebSphere Voice Response function that uses the terms. For example, one group is called “Administrator Profile Terms.” This group includes the text that you see in the windows that you use to work with administrator profiles. Another is called “Custom Server Terms.” This group includes the text that you see in the windows that you use to work with custom servers.

Each term is also labeled with an internal tag. That tag enables WebSphere Voice Response to identify the term after you have translated it.

**Introducing help text**

The system includes help text for all the WebSphere Voice Response windows. This is the text that you see when you click Help —> On Window, or when you display the information in the help index.

The help text for each window is in a separate file. These files are cataloged by window name and type. The types are:

- Main
- Work
- Field

The help text in a main file describes how to use a main window. WebSphere Voice Response displays a main window when you click an action from one of the menus on the Welcome window. The help text in a work file describes how to use a work window. A work window is any window that is not a main window. The help text in a field file describes an action on a menu.

To format help information for display, WebSphere Voice Response uses a set of Generalized Markup Language (GML) tags. Each tag controls a different format characteristic. For example, one tag specifies that text be displayed in bold-faced type. Another tag specifies that text be displayed as a bulleted list. The tags, and what each tag controls, are listed in Table 17.

<table>
<thead>
<tr>
<th>Tags</th>
<th>Usage</th>
<th>Example</th>
<th>Formatted Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>:h1.</td>
<td>Headings</td>
<td>:h1.Heading Level 1</td>
<td>Heading Level 1</td>
</tr>
<tr>
<td>:h2.</td>
<td></td>
<td>:h2.Heading Level 2</td>
<td>Heading Level 2</td>
</tr>
</tbody>
</table>
Table 17. GML tags for formatting help information (continued)

<table>
<thead>
<tr>
<th>Tags</th>
<th>Usage</th>
<th>Example</th>
<th>Formatted Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>:p.</td>
<td>Paragraph</td>
<td>:p. Put this tag at the start of a paragraph. :p. The tag leaves a blank line.</td>
<td>Put this tag at the start of a paragraph. The tag leaves a blank line.</td>
</tr>
<tr>
<td>:hp1.</td>
<td>Highlighted phrases</td>
<td>:hp1. Italics:ehp1. introduce new terms. Boldface is used for :hp2. window items:ehp2. Note the double period when a tag also ends a sentence.</td>
<td>Italics introduce new terms. Boldface is used for window items. Note the double period when a tag also ends a sentence.</td>
</tr>
<tr>
<td>:hp2.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>:xph.</td>
<td>Examples</td>
<td>Window :xph.input/output:exph. is shown in example font.</td>
<td>Window input/output is shown in example font. Program example line 1 Program example line 2</td>
</tr>
<tr>
<td>:xmp.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>:ol.</td>
<td>Numbered list</td>
<td></td>
<td></td>
</tr>
<tr>
<td>:li.</td>
<td>Item in either list</td>
<td></td>
<td></td>
</tr>
<tr>
<td>:dl.</td>
<td>Definition list</td>
<td>:p. This window displays the following fields: :dl. :dt. Relation :dd. Specifies ... :dt. Record :dd. Identifies ... :edl.</td>
<td>This window displays the following fields: Relation Specifies ... Record Identifies ...</td>
</tr>
<tr>
<td>:edl.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>:dt.</td>
<td>Term Definition</td>
<td></td>
<td></td>
</tr>
<tr>
<td>:dd.</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Using WebSphere Voice Response to translate display text

To translate display text into another language, use this procedure and the editor that is provided with WebSphere Voice Response. To translate help text, use the procedure that is described in “Using WebSphere Voice Response to translate help text” on page 171. To export display text to an ASCII file that you can edit with a different editor, use the procedure that is described in “Copying display text to an ASCII file on the hard disk” on page 173.

Some PTFs provide new display text. If you notice unexpected text, such as oamui.enable_button, after applying PTFs, you must translate the new display text.
Procedure

1. From the Welcome window, select Configuration —> Languages.

2. Displaying text to translate: Open the language into which you want to translate display text.
   The system displays the Language window.

3. Click Options.

4. Click Language String Editor.
   The system displays the text that is in the first group of terms in the system.

5. Displaying different text to translate: Click String Group....
   The system lists all the groups of terms in alphanumeric order, starting with groups whose names begin with numbers.
6. Click the group that includes the text that you want to translate.

7. Click OK.

   The system lists the text for all terms that are in the selected group. The terms are listed in alphanumeric order, starting with terms that begin with a number.

8. Translating the text: Click the term that you want to translate.

   The system copies the term to the edit field that is at the bottom of the window. The field is relabeled with the internal tag that WebSphere Voice Response uses to recognize the term.

9. Type a translation of the term in the edit field.

10. Translating more text: To save the translation and translate more terms, repeat Steps 5 on page 169 through 8 or Steps 7 and 8 until you are finished.

11. Click OK.

   The system displays the Language window.

12. Saving and Closing: Click File —> Save.

13. Click File —> Close.

   The translated terms are saved in the database for the selected language.
Using WebSphere Voice Response to translate help text

To translate help text into another language, use this procedure and the editor that is provided with WebSphere Voice Response.

Help text is sometimes updated with WebSphere Voice Response PTFs. If you install a PTF after translating your help text, perform this procedure again.

Unless you have already translated the display text, window titles are displayed in the language that is used as the source for the window text. To translate the window titles, use the procedure that is given in “Using WebSphere Voice Response to translate display text” on page 168 or in “Using another editor to translate display text” on page 172.

The help text editor does not include an automatic line wrap feature. When you are translating text and the line is going to extend beyond the edge of the window, press <Enter> to start a new line. <Enter> does not create a "hard" carriage return (start a new paragraph). When WebSphere Voice Response formats the help text, it starts new lines that are related to the window size, not to where you pressed <Enter>. It starts new paragraphs when it finds a :p. tag (described in Table 17 on page 167). Note that, if you print help text from the help text editor, the print program does not include an automatic line wrap feature; therefore, it truncates each line of the printed copy after 80 columns.

Procedure
1. From the Welcome window, select Configuration —> Help Editor.
2. Selecting the language: Click the language into which you are translating the help text.
   The system lists the titles of the main windows.
3. Opening a main file: To translate help text in a main file, click the file.
4. Click Main and Open the file.
   The system displays the file.
5. Opening a work file: To translate help text for a work window, click the name of the main window from which you access the work window.
   The system adds the titles of the work windows to the display.
6. Click the file.
7. Click Work and Open the file.
   The system displays the file.
8. Opening a field file: To translate help text in a field file, click the name of the main window from which you access the work window on which the menu is located.
The system displays the titles of the work windows that are accessible from the main window.

9. Click the name of the work window on which the menu is located.
   The system lists all menu actions that are accessible from the work window.

10. Click the file.

11. Click Field and Open the file.
    The system displays the file.

12. **Translating the help text:** To translate by editing the existing text, make your changes to the text that is on the right (the **Current Work**).

13. To translate by rewriting the existing text, click Clear and type in the new text.

14. To discard any changes you have made and copy the existing text to start again, click Copy Original.

15. **Checking the format:** To display your changed text as it will appear when it is formatted, click Options —> Format Current.

16. To display the original format, click Options —> Format Original.

17. **Saving the translated text:** Click File —> Save.
    The translated version is now available when windows are displayed in the new language.

18. **Translating additional text:** Click File —> Close.
    The system displays the list of window titles.

19. Open another file by repeating Steps 3 on page 171 and 4 on page 171 through 11 and then repeating the rest of this procedure.

When you are finished, Close the Help Editor window.

**Using another editor to translate display text**

WebSphere Voice Response can copy display text to a flat ASCII file on the hard disk in the pSeries computer. To translate the text, you can then use an editor other than the editor that is provided with WebSphere Voice Response. When you are finished, you can use WebSphere Voice Response to copy the translated file back from the hard disk into the WebSphere Voice Response database.

When you copy the translated information back into WebSphere Voice Response, the system merges the translated information with information that is in the language database. Therefore, if you accidentally deleted a term when you translated the information, the system still runs. Of course, the term that you deleted is displayed in the original language.
Copying display text to an ASCII file on the hard disk

Use this procedure to copy display text to an ASCII file on the pSeries computer hard disk so that you can translate it by using another editor.

To translate display text using the editor that is provided with WebSphere Voice Response, use the procedure that is described in "Using WebSphere Voice Response to translate display text" on page 168.

When you translate display text, ensure you do not change the internal tag that identifies each term. If you do, WebSphere Voice Response no longer recognizes the term. The term that is displayed is not be translated, because WebSphere Voice Response cannot find the translated term.

Procedure

1. From the Welcome window, select Configuration —> Languages.
2. Displaying languages by type: To list all languages for which display text exists, click View —> Window Text Languages.
   The system displays all languages that are defined as Window Text or Window Text and Voice.

3. Selecting the language: Open the language into which the display text is to be translated.
4. The system displays the Language window.
5. Click Options.
6. Click Export Language Strings.
   The system displays the file search display.
7. Identifying a file in which to store the text: If you are copying to an existing file, type the file path name in the Selection field, or use the file search display to find the directory and file and click the file name.
   The system displays the path name in the Selection field.
8. Creating a file in which to store the text: To create a file in which to store the exported text type the path name that identifies the new file in the Selection field, or use the file search display to find a directory that is not used by WebSphere Voice Response (for example, your /home directory) and add the new file name to the end of the path name displayed in the Selection field.
Click OK.

The system copies the text and displays the Language window.

**Copying display text from the hard disk**

Use this procedure to copy back display text that you have finished translating. To copy the text to translate it, use the procedure that is described in “Copying display text to an ASCII file on the hard disk” on page 173.

When you copy translated text back into WebSphere Voice Response, WebSphere Voice Response checks the file to ensure that it is one you previously copied out, and is therefore in the correct format. WebSphere Voice Response then merges the text with any text that already exists in the database for that language. You cannot copy display text back into the English database.

**Procedure**

1. From the Welcome window, select **Configuration —> Languages**.
2. **Selecting the language**: Open the language into which the display text has been translated.
   
The system displays the Language window.
3. Click **Options**.
4. Click **Import Language Strings**.
   
The system displays the file search display.
5. **Identifying the file in which the text is stored**: Type the path name that identifies the file on the hard disk in the **Selection** field or use the file search display to find the file and click the file name.
6. The system displays the path name in the **Selection** field.
7. Click **OK**.
   
The system imports the text and displays the Language window.
8. **Save** the translated text.
If you Close the Languages window without saving, the system prompts you to save or discard the imported text.

Moving translated text to a different WebSphere Voice Response system

You can copy translated text from one WebSphere Voice Response system to another. For example, if you have more than one system, and both systems should be usable in both English and French, it is not necessary to translate the window text twice. Instead, you can use the dtexport and dtimport utilities to copy translated window text from one system to the other system.

About the export utility

The dtexport utility copies a file (or group of files) from WebSphere Voice Response to tape, diskette, or the hard disk in the pSeries computer. WebSphere Voice Response writes to the target file without checking to see whether it is empty.

If you are exporting to tape or diskette, WebSphere Voice Response gives the exported file the same name as the tape or diskette drive. If you are exporting to the hard disk, you can tell the system to store the text in an existing file (assuming the file is empty or that you no longer need the contents), or you can create a new file as part of the export process.

About the import utility

The dtimport utility copies a file (or group of files) from tape, diskette, or the hard disk in the pSeries computer back to the WebSphere Voice Response file system. When the file has been imported, you must also install it in the database before WebSphere Voice Response can use it.

Any file that you import must have been previously exported. Otherwise, it is in the wrong format and WebSphere Voice Response does not import it.

Moving window text

To move translated window text, do the following:

1. Define the new language and create the new language database on the target system.

2. Export the translated text to tape or diskette. Use the instructions that are given in the WebSphere Voice Response for AIX: Managing and Monitoring the System book to export the text.

3. Import the translated text from the tape or diskette to the target database and install it. Use the instructions that are given in the WebSphere Voice Response for AIX: Managing and Monitoring the System book to import and install the text.

Text that you import by using the import utility completely replaces text in the new database.
Using translated system prompts

For information about using system prompts that have been translated into languages other than U.S. English, including those delivered with WebSphere Voice Response, see the WebSphere Voice Response for AIX: Managing and Monitoring the System book.

Changing the technical difficulties message

The technical difficulties message is a prerecorded message that is supplied with WebSphere Voice Response. It says "We are experiencing technical difficulties. Please hang up and try again later." to callers when WebSphere Voice Response answers an incoming call and cannot process it. This can be for one of the following reasons:

- WebSphere Voice Response has lost its connection to DB2 or to the file systems that contain your voice and customer applications.
- WebSphere Voice Response is configured to go off hook before attempting to get call information (see “EDL Call Information After Off Hook” on page 305). After playing the message, WebSphere Voice Response hangs up.
- WebSphere Voice Response cannot find an application to handle the call, such as the supplied Incoming_Call state table.
- A state table is not valid. The state table might be not valid in memory only, or the state table on disk might also be not valid.

As System Administrator you might want to change the technical difficulties message so that callers to hear a message that is specific to your business or in a different language.

The technical difficulties message is loaded by WebSphere Voice Response so that it can be played although the caller might have problems accessing the database.

The technical difficulties message can be played:

- Directly when a channel process (CHP) dies.
- At the request of the channel process, when the channel process cannot access voice data. For this problem, an error is put into the error log.
- When an outage of a WebSphere Voice Response file system or database occurs.

How to create a new technical difficulties message

Use the following procedure to record a new voice segment that is to be used as the technical difficulties message, and to use the new message.

1. Record the new message as a voice segment. You can use the supplied state table Record_Uncomp for this.
2. Use the Voice Segment Editor to export the segment to a file:
   - **Welcome window** —> **Applications** —> **Voice Segments** then choose Language, Voice Directory and Voice Segment then **Utilities** —> **Export**
   - Leave defaults and choose filename and click OK.

3. **Replace the message:** Log in as **dtuser**.

4. If the system displays the Login menu:
   - **WebSphere Voice Response User Login**
     - 1) Start WebSphere Voice Response Processes
     - 2) Do Not Start WebSphere Voice Response
     - Enter choice (or <ENTER> for option list)
   - Select option 2.

5. Type the following command and press Enter:
   - `su root`

6. Run the script `$VAETOOLS/TechDiff` using the following parameters:
   - `-d` sets the maximum decibel level of the message. A good starting point for this value is -11. If the message is quiet, you can make it louder by using a value of -14.
   - `-f` sets the voice file (full path)
   - For example:
     - `TechDiff -d-11 -f/home/dtuser/TechDiff.comp`
   - When using the `-d` and `-f` parameters both must be present or both absent.

   **Note:** If you are using a PCI machine and the technical difficulties message is not available, error 27010 (pack enablement failed) is generated and the file name is shown in the explanation.

7. **Querying the current setting:** You can query the setting of the technical difficulties message by using:
   - `TechDiff -q`
   - The message can be reset to the supplied default by using:
   - `TechDiff -r`

   **When the new message is available:** The new technical difficulties message is loaded when WebSphere Voice Response is restarted.
Appendix A. System parameters

This appendix has two sections:

- "System parameter groups" describes the groups into which the WebSphere Voice Response system parameters are divided. For each group, this section introduces the group and lists the parameters in that group.
- "System parameters reference" on page 195 is an alphabetic listing of all the system parameters. This section gives a full description of each parameter.

Note: Any adjustments that might affect compliance with telecommunications authority regulations are to be made only by authorized personnel who are familiar with these requirements.

Related Information:
- "Introducing the system parameters" on page 12
- Appendix B, "System parameter templates," on page 575

System parameter groups

Parameters in the following groups are defined once, and apply to the whole system:

- "Application server interface parameter group" on page 180, which defines the characteristics of all voice applications.
- "CPU monitor parameter group" on page 185, which specifies the operation of the CPU monitor.
- "Exchange data link parameter group" on page 185, which defines the characteristics of the exchange data link (optional).
- "General parameter group" on page 186, which specifies some aspects of WebSphere Voice Response operation, including log file archiving.
- "ISDN signaling parameter group" on page 187, which defines values for use by the ISDN subsystem (optional).

In addition, you can define:

- Up to 16 E1 or T1 trunks, using the parameters in the "Trunk interface parameter group" on page 191.
- Up to 16 signaling types, using the parameters in the "Signaling type parameter group" on page 189 (channel associated signaling only).
- Up to 16 channel groups, using the parameters in the "Channel group parameter group" on page 184. Each channel group can include up to 120
channels. Each channel group definition specifies the signaling type to which it belongs (channel associated signaling only) or a signaling process type.

- Up to 480 E1 channels or 384 T1 channels, using the parameters in the Channel parameter group on page 183. Each channel definition specifies the channel group to which it belongs. Each channel inherits the characteristics of:
  1. The trunk to which it is assigned (see the WebSphere Voice Response for AIX: Installation book)
  2. The channel group to which it belongs
  3. The signaling type to which the channel group belongs.
- Up to 25 call progress tones, using the parameters in the Call progress tones parameter group on page 182 (for outbound dialing only).
- Characters or symbols to be mapped to 16 keys on the telephone key pad, using the parameters in the Key signals parameter group on page 188 (optional).

All of these parameters have default values, which are listed in the System parameters reference on page 195. You need to change the values only if they are not appropriate to your system. For information about how to set these values, see “Setting the value of a system parameter” on page 13.

Not all parameter values take effect immediately. In some parameter groups, the new values take effect when you restart WebSphere Voice Response (see “When do new values take effect?” on page 16) and in others, the new values take effect when you disable and then enable the packs (see the WebSphere Voice Response for AIX: Managing and Monitoring the System book).

The following sections describe the parameter groups, tell you what to do to make new parameter values take effect, and list the parameters in each group.

**Application server interface parameter group**

**Purpose**

The parameters in this group define how WebSphere Voice Response voice applications interact with callers.

**Scope**

These parameters must be defined once. The definitions apply to all the applications in the system. You can override some of the values for a particular application by using system variables (see the WebSphere Voice Response for AIX: Application Development using State Tables book).

To make new parameter values effective, restart WebSphere Voice Response.
Parameters

- 3270 Mode (page 195)
- Alarms - Make All Alertable (page 203)
- Alarms - Send to AIX Error Log (page 204)
- Audio Name Compression Type (page 212)
- Check Voice Messages Time - Alert (ms) (page 242)
- Check Voice Messages Time - Max Allowable (ms) (page 243)
- Check Voice Messages Time - Recovered (ms) (page 244)
- CHPM Socket Port Number (page 246)
- CHP Performance Metrics - Expiry Time (mins) (page 247)
- CHP Performance Metrics - Weighting of Old Average (page 248)
- Database Availability Check Timeout (page 259)
- Default System Prompt Directory Name (page 267)
- DTTA Loading Clear Threshold (%) (page 295)
- DTTA Loading Warning Threshold (%) (page 296)
- DTTA Interrupt Separation Clear Threshold (ms) (page 293)
- DTTA Interrupt Separation Warning Threshold (ms) (page 294)
- EDL Call Information After Off Hook (page 305)
- EDL Message Info Age Limit (Seconds) (page 308)
- EDL Message Info Time Out (Seconds) (page 309)
- Enter Key (page 313)
- Errorlog Wrap Threshold (recs) (page 315)
- Extra Channel Process (page 316)
- File Availability Check Timeout (page 317)
- Forward Key (page 318)
- Low Channel Process Clear Threshold (page 369)
- Low Channel Process Warning Threshold (page 370)
- Maximum Cached Buffers (page 374)
- Maximum Dial Tone Wait (Seconds) (page 375)
- Maximum MPN Digits (page 376)
- Maximum Ring Time (Seconds) (page 380)
- Maximum Ring Wait (Seconds) (page 382)
- Music Automatic Fade Before Actions (page 394)
- Music Automatic Fade Time Default (ms) (page 395)
- Music Volume Ceiling Default (dBm) (page 397)
- Normal Play/Record Max Data (KBytes) (page 403)
- Number of 3270 Exec Processes to Spawn (page 404)
- Number of Non Swap State Tables (page 406)
- Number of Pool Buffers (page 407)
- Password Minimum Length (page 419)
- Pause Key (page 420)
- Play Latency - Max Allowable (ms) (page 422)
- Play Latency - Recovered (ms) (page 423)
- Play Latency Time - Alert (ms) (page 424)
- Play Skip (Seconds) (page 425)
- Profile Retrieval Time - Alert (ms) (page 427)
Call progress tones parameter group

Purpose

The parameters in this group define the characteristics of each tone (dial tone, busy tone, and so on). These tones are used when making an outgoing call, or transferring a call, to provide feedback about the status of the call to the state table.

Several predefined templates are available. Select the correct template for your country or region and switch, and copy it to one of the numbered tone groups. Then specify that tone group in the Tone Group parameter in the Channel Group parameter group. If you need to change the tone definitions, make the changes in the numbered tone group. For general information about call progress tones, see "Setting call progress tone parameters for outbound"
Scope

The call progress tone parameters must be defined once for each tone that is provided by the switch.

To make new parameter values effective, disable then enable the packs.

Parameters

- Frequency 1 Maximum (Hz) (page 319)
- Frequency 1 Minimum (Hz) (page 320)
- Frequency 2 Maximum (Hz) (page 321)
- Frequency 2 Minimum (Hz) (page 322)
- Frequency 3 Maximum (Hz) (page 323)
- Frequency 3 Minimum (Hz) (page 324)
- Level 1 Maximum (dBm) (page 361)
- Level 1 Minimum (dBm) (page 362)
- Level 2 Maximum (dBm) (page 363)
- Level 2 Minimum (dBm) (page 364)
- Level 3 Maximum (dBm) (page 365)
- Level 3 Minimum (dBm) (page 366)
- Time Off 1 Maximum (0.001 Seconds) (page 529)
- Time Off 1 Minimum (0.001 Seconds) (page 530)
- Time Off 2 Maximum (0.001 Seconds) (page 531)
- Time Off 2 Minimum (0.001 Seconds) (page 532)
- Time Off 3 Maximum (0.001 Seconds) (page 533)
- Time Off 3 Minimum (0.001 Seconds) (page 534)
- Time On 1 Maximum (0.001 Seconds) (page 535)
- Time On 1 Minimum (0.001 Seconds) (page 536)
- Time On 2 Maximum (0.001 Seconds) (page 537)
- Time On 2 Minimum (0.001 Seconds) (page 538)
- Time On 3 Maximum (0.001 Seconds) (page 539)
- Time On 3 Minimum (0.001 Seconds) (page 540)
- Tone Label (page 542)
- Tone Type (page 543)

Channel parameter group

Purpose

The parameters in this group define the telephony channels to which WebSphere Voice Response is connected. The parameters specify the channel group to which the channel belongs, and two identifiers for the channel: the EDL message information line identifier on the exchange data link (if used), and the phone number for incoming calls. Other characteristics of the channel...
are defined by the channel group parameters.

**Scope**

The channel parameters must be defined once for each channel. Select the channel that you want to define and open it.

To make new parameter values effective, disable then enable the packs to which the channels are connected.

**Parameters**

- Channel Group (page 241)
- Message Info Line Identifier (page 388)
- Phone Number (page 421)

**Channel group parameter group**

**Purpose**

The parameters in this group define the telephony channels to which WebSphere Voice Response is connected. The parameters specify trunk protocols and other signaling criteria for all the channels included in a channel group.

**Scope**

The channel group parameters must be defined once for each channel group. You can define up to 16 channel groups. The parameter values then apply to all the channels in the group. The channels in a group can be connected to more than one trunk. (The Channel Group parameter in the Channel parameter group specifies the group to which the channel belongs.)

Select the channel group that you want to define and open it.

To make new parameter values effective, disable then enable the packs to which the channels in the channel group are connected.

**Parameters**

- Area Code (page 211)
- Call Information Type (page 225)
- CAS - Allow Alternate Hangup (page 237)
- Connect Voice Channel Before Answer (page 252)
- Dial Tone Detection (page 272)
- DID Start Type (page 275)
- Direction (page 276)
- E1 CAS Protocol (page 298)
- E&M Start Type (page 297)
• FXS Start Type (page 325)
• Incoming Address Signaling Type (page 344)
• Outgoing Address Signaling Type (page 415)
• Signaling Process Type (page 495)
• Signaling Type (page 498)
• T1 CAS Protocol (page 520)
• Tone Group (page 541)
• UK Tie/DDI Start Type (page 556)
• Voice Interrupt Detection Level (dBm) (page 564)
• Voice Interrupt Detection On Time (ms) (page 566)
• Voice Interrupt Detection Off Time (ms) (page 565)

CPU monitor parameter group

Purpose

The parameters in this group define the operation of the CPU monitor.

The monitor uses the percentage of CPU idle time to determine the load on the system. When the CPU is idle only for very short times, the system is heavily loaded and running very slowly. (For more information on how the CPU monitor operates, see the WebSphere Voice Response for AIX: Managing and Monitoring the System book.)

Scope

These parameters must be defined once.

To make new parameter values effective, restart WebSphere Voice Response.

Parameters

• CPU Clear (page 257)
• CPU Warning Threshold (page 258).

Exchange data link parameter group

Purpose

The parameters in this group define the exchange data link (if used). The exchange data link carries SMSI, SMDI, VMS, ACL, or other out-of-band information. To implement the exchange data link, you must also set the Message Info Line Identifier parameter in the Channel parameters group for each channel.

Scope

These parameters must be defined once, if you are using an exchange data link.
To make new parameter values effective, restart WebSphere Voice Response.

**Parameters**

- Called Number Character to Strip (page 228)
- Called Number Length (page 229)
- Called Number Length (Minimum) (page 230)
- Called Number Stripping (page 231)
- Calling Number Character to Strip (page 232)
- Calling Number Length (page 233)
- Calling Number Length (Minimum) (page 234)
- Calling Number Stripping (page 235)
- EDL Communication Port (page 306)
- EDL Data Rate (bits/sec) (page 307)
- EDL Parity (page 310)
- EDL Switch Type (page 311)
- Interval for Checking MWI Status (s) (page 348)
- Line Identifier Number Length (page 368)
- MWI Automatically Set (page 398)
- MWI Number Length (page 399)
- MWI Number Padding (page 400)
- MWI Number Padding Character (page 401)
- Number of Nak Retries (page 405)
- System Number (page 517)

**General parameter group**

**Purpose**

The parameters in this group specify some aspects of WebSphere Voice Response operation.

**Scope**

These parameters must be defined once.

To make new parameter values effective, restart WebSphere Voice Response.

**Parameters**

- Allow incoming numbers with Presentation Restricted (page 207)
- Buffer Pool Address (page 216)
- Call Detail Record Logging (page 224)
- Control Memory Address (page 255)
- Country or Region (page 256)
- DBIM Time Out (page 260)
- Default Diskette Drive (page 265)
- Default Tape Drive (page 268)
- Error Table Address (page 314)
ISDN signaling parameter group

Purpose

The parameters in this group define values for use by the ISDN subsystem.

Scope

These parameters must be defined once if you are using ISDN signaling.

To make new parameter values effective, stop the ISDN signaling process and disable all ISDN trunks, then re-start and re-enable.
Parameters

- Alert level (page 206)
- B-Channel Service Message Support (page 214)
- Calling Party Number — MWI Identification (page 236)
- D-Channel Service Message Support (page 261)
- L2 - Link Handshake Timer T203 (ms) (page 354)
- L2 - Link Release Timer T200 (ms) (page 355)
- L3 - T309 Support (ms) (page 356)
- L4 - Called/Calling Party Numbering Plan (page 357)
- L4 - Called/Calling Party Numbering Type (page 358)
- L4 - Facility Timeout (s) (page 359)
- L4 - Facility Transfer Completion Timeout (s) (page 360)
- Maintenance Message Protocol Discriminator (page 371)
- MWI Trunk (page 387)
- Redial Limitation - Failed List Capacity (page 445)
- Redial Limitation - Maximum Consecutive Failures (page 446)
- Redial Limitation - Significant Digits (page 447)
- Redial Limitation - Timeout (page 448)
- Send RESTART on Channel Enable (page 487)

Key signals parameter group

Purpose

Each of 16 keys on the telephone keypad can have characters or symbols mapped to them. This assignment enables WebSphere Voice Response to interpret DTMF signals as text. (For more information on how voice applications distinguish between different letters, see the GetText action in the WebSphere Voice Response for AIX: Application Development using State Tables book.)

Scope

The Map parameter (the only parameter in this group) must be defined once for each key on the telephone keypad.

To make new parameter values effective, restart WebSphere Voice Response.

Parameters

Map is the only parameter in this group (see “Map” on page 372).

There is a predefined value for each key. The predefined values are:
For example, to move Q and Z from the 1 key to the 0 key:

1. Select Key Signals in the System Configuration window.
2. Select the button marked 1.
3. Select Map.
4. Select the New Value and delete it.
5. Select OK.
6. Close the Key Signals / 1 window.
7. Select the button marked 0.
8. Select Map.
9. Select the New Value and overtype it with QZ0.
10. Select OK.
11. Close the Key Signals / 0 window.
12. Close the Key Signals window and then Save your changes.

### Signaling type parameter group

#### Purpose

The parameters in this group define the timing and signaling values used by all the channels in a channel group that use channel associated signaling (CAS). The values are assigned to a channel group by specifying the Signaling parameter for the channel group.

#### Scope

The signaling type parameters must be defined once for each signaling type. The values then apply to all channels in all channel groups that specify that signaling type. You can define up to 16 signaling types.

To make new parameter values effective, disable then enable the packs to which channels that use the signaling group are attached.

#### Templates

Some predefined signaling types are supplied as templates (see Appendix B, “System parameter templates,” on page 575). You cannot edit the values in the
templates. You must copy the templates and paste them onto numbered signaling types (1 through 16).

**Parameters**
- Answer Delay Time (ms) (page 206)
- Blocking Action (page 215)
- Cadence Energy Maximum (dBm) (page 217)
- Cadence Energy Minimum (dBm) (page 218)
- Cadence Off Time Maximum (ms) (page 219)
- Cadence Off Time Minimum (ms) (page 220)
- Cadence On Time Maximum (ms) (page 221)
- Cadence On Time Minimum (ms) (page 222)
- Cadence Silence Maximum (dBm) (page 223)
- CO Acknowledgment (ms) (page 249)
- CO Off Hook (ms) (page 250)
- CO On Hook (ms) (page 251)
- Constant Energy Maximum (dBm) (page 253)
- Constant Energy Minimum (dBm) (page 254)
- Delay Start Delay (ms) (page 269)
- Delay Start Duration (ms) (page 270)
- Dial Pause (ms) (page 271)
- Dial Tone Qualify Time (ms) (page 272)
- Dial Tone Timeout (ms) (page 273)
- Glare Detection Time (ms) (page 332)
- Ground Flash (ms) (page 334)
- Hang Up Detection (page 336)
- Hook Flash (ms) (page 337)
- Incoming Address Register Type (page 343)
- Incoming Guard Time (ms) (page 345)
- No Answer Warning (ms) (page 401)
- Outgoing Address Register Type (page 414)
- Outgoing Guard Time (ms) (page 416)
- Reconnect Call Feature Code (page 458)
- Reconnect Call Request Signal (page 440)
- Register Length (page 454)
- Ringing Off Maximum (ms) (page 460)
- Ringing Off Minimum (ms) (page 461)
- Ringing On Maximum (ms) (page 462)
- Ringing On Minimum (ms) (page 463)
- Seize Acknowledgment Timeout (ms) (page 485)
- T1 CAS Signaling Format (page 521)
- Transfer Call Feature Code (page 544)
- Transfer Call Request Signal (page 546)
- Wink Start Delay (ms) (page 572)
- Wink Start Duration (ms) (page 573)
Trunk interface parameter group

Purpose

The parameters in this group define the timing and signaling values for a trunk. The field-level parameters in this group define signaling levels, transmission speeds, and cadences for the trunk.

Scope

The trunk interface parameters must be defined once for each pack. The values then apply to all channels assigned to that trunk. Up to 16 T1 or E1 trunk interfaces can exist.

To make new parameter values effective, disable then enable the pack.

Templates

Some predefined trunk interfaces are supplied as templates (see “Trunk interface templates” on page 585). You cannot edit the values in the templates. You must copy the templates and paste them onto numbered trunk interfaces (1, 2, 3, and so on).

Parameters

- Answer Detect Threshold (dBm) (page 209)
- Answer Detect Time (ms) (page 210)
- Backup Time and Erase after DTMF (Interrupts) (page 213)
- CCS Clustered mbufs in Receive Pool (page 238)
- CCS mbufs in Receive Pool (page 239)
- CCS Signaling Link Mode (page 240)
- DP Receive Maximum Break (ms) (page 279)
- DP Receive Maximum Make (ms) (page 280)
- DP Receive Minimum Break (ms) (page 281)
- DP Receive Minimum Make (ms) (page 282)
- DP Transmit Break (ms) (page 283)
- DP Transmit Speed (pulse/sec) (page 284)
- DTMF Algorithm Variant (page 285)
- DTMF Maximum Receive Level (dBm) (page 286)
- DTMF Minimum Receive Level (dBm) (page 287)
- DTMF Transmit Level, Low Frequency (dBm) (page 289)
- DTMF Transmit Level Twist (dBm) (page 290)
- DTMF Transmit On (ms) (page 291)
- DTMF Transmit Speed (digits/sec) (page 292)
- E1 Framing Mode (page 299)
- E1 Hit Filter (2 ms) (page 300)
- E1 Timeslot 0 Word (page 301)
- E1 Timeslot 16 Word (page 302)
This section lists all the Voice over IP system parameters. It gives the following information about each. All system parameters, for both DTEA and DTNA media are contained within the same system parameters group (VoIP DTEA and DTNA Media). Not all parameters apply to both DTEA and DTNA media. See each parameter description for applicability.
Scope

The VoIP DTEA and DTNA Media parameters must be defined once. DTEA Media parameters are effective for all DTEA packs (EPACKS) and DTNA Media parameters and are effective for all DTNA packs (NPACKS).

To make new parameter values effective, disable and enable all DTEA or DTNA packs (as appropriate).

Parameters

- 1st Codec Preference (page 196)
- 2nd Codec Preference (page 197)
- 3rd Codec Preference (page 199)
- 4th Codec Preference (page 200)
- DTMF Transmission Method (page 288)
- Enable Echo Cancellation (page 312)
- G711 Voice activity det/comfort noise gen (page 326)
- G711 Packet Voice Interval (ms) (page 327)
- G729 Voice activity det/comfort noise gen (page 328)
- G729 Packet Voice Interval (ms) (page 329)
- G723 Voice activity det/comfort noise gen (page 330)
- G723 Data Transfer Rate (page 331)
- G723 Packet Voice Interval (ms) (page 332)
- Inbound DTMF Method Override (page 342)
- Outbound DTMF Method Override (page 412)
- Override DTNA RTP Transport IP Address (page 466)
- RTCP Enable Sender Report (page 464)
- RTCP Sender Report Interval (page 465)
- RTP Base Port Number (page 467)
- RTP IP TOS Byte (TOS) (page 468)
- RTP IP Time to Live (TTL) (page 469)
- RTP Security Negotiation (page 470)

VoIP Media-Adapters parameter group

Purpose

The parameters in this group define how the DTEA cards connect to the IP network.

Scope

The media adapter parameters must be defined once for each installed DTEA card, with a maximum of four cards.

Select the media adapter that you want to define and open it.
To make the parameter values effective, restart WebSphere Voice Response.

**Parameters**
- IP Address (page 349)
- Subnet Mask (page 309)
- Default RTP Router (page 266)

**VoIP SIP Signaling parameter group**

**Purpose**

The parameters in this group define values for use by the SIP signaling process.

**Scope**

These parameters must be defined once if you are using SIP signaling.

The parameters defined here are used for all SIP channels.

To make new parameter values effective, disable and enable all DTEA packs

**Parameters**
- Accept Inbound Transfer Requests (page 201)
- Add Host Name To User Agents? (page 202)
- Call Signaling Port (page 227)
- CHP available call reject threshold (page 245)
- Default CLID for Incoming VoIP Calls (page 262)
- Default Destination Port (page 264)
- Default Destination URI (page 264)
- DNSSRV Server Address (page 277)
- DNSSRV Server Port (page 278)
- E164 Prefixes to Strip (page 303)
- Ignore replaces option for Attended Transfer (page 338)
- Inbound Call Channel Allocation Method (page 340)
- Message Header Format (page 386)
- Organization Name (page 411)
- Outbound SIP INFO (page 413)
- Override SIP Transport IP Address (page 417)
- Proxy Address (page 432)
- Proxy Mode (page 433)
- Proxy Port (page 435)
- Register Addresses on Startup (page 451)
- Register Default Timeout (Minutes) (page 452)
- Register Default User Agent (page 453)
- RFC3264 Media on-hold method (page 459)
- Secure SIP Enabled (page 484)
System parameters reference

This section lists all the system parameters in alphabetic sequence. It gives the following information about each:

**Information structure**

**Parameter group**

The group to which the parameter belongs, and the name on the button in the System Configuration window that gives access to the parameter values.

**Applicability**

The VoIP media to which the parameter applies (DTEA and or DTNA).

**Access level**

Admin or Field. For more information, see “Access control” on page 6.

**Possible values**

The range, or a list of possible values, to which you can set the parameter.

**Defaults**

The value to which the parameter at WebSphere Voice Response installation.

**Explanation**

A brief explanation of the function of the parameter and, if necessary, the meaning of the values.
1st Codec Preference

Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA and DTNA

Access level
Admin

Possible values
G711 ALAW (64Kb/s)
G711 MULAW (64Kb/s)
G723
G729
iLBC 20
iLBC 30

Defaults
G711 ALAW (64Kb/s)

Explanation
The codec that will be the first choice when negotiating media streaming. For DTEA, this value can be set to any of the above values except iLBC 20, or iLBC 30. For DTNA, this value should only be set to None, G711, A law, μ law, iLBC 20, or iLBC 30 otherwise an error message will be issued when the trunk is enabled.
2nd Codec Preference

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA and DTNA

Access level

Admin

Possible values

None
G711 ALAW (64Kb/s)
G711 MULAW (64Kb/s)
G723
G729
iLBC  20
iLBC  30

Defaults

G711 ALAW (64Kb/s)

Explanation

The codec that will be the second choice when negotiating media streaming. If this is set to None all subsequent codecs will be ignored. For DTEA, this value can be set to any of the above values except iLBC 20, or iLBC 30. For DTEA, this value can be set to any of the above values. For DTNA, this value should only be set to None, G711, A law, µ law, iLBC 20, or iLBC 30 otherwise an error message will be issued when the trunk is enabled.
3270 Mode

Parameter group

Application Server Interface

Access level

Admin

Possible values

Real Mode
Virtual Mode

Defaults

Real Mode

Explanation

Specifies how 3270 servers are to run. If WebSphere Voice Response is attached to a 3270 database host with which 3270 servers communicate, select Real Mode. Virtual Mode is useful for testing and demonstrating a 3270 server when WebSphere Voice Response is not attached to a 3270 host.
3rd Codec Preference

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA only

Access level

Admin

Possible values

None
G711 ALAW (64Kb/s)
G711 MULAW (64Kb/s)
G723
G729
iLBC 20
iLBC 30

Defaults

None

Explanation

The codec that will be the third choice when negotiating media streaming. If this is set to None all subsequent codecs will be ignored. For DTEA, this value can be set to any of the above values except iLBC 20, or iLBC 30. For DTNA, this value should only be set to None, G711, A law, µ law, iLBC 20, or iLBC 30 otherwise an error message will be issued when the trunk is enabled.
4th Codec Preference
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
None
G711 ALAW (64Kb/s)
G711 MULAW (64Kb/s)
G723
G729
iLBC 20
iLBC 30

Defaults
None

Explanation
The codec that will be the fourth choice when negotiating media streaming. For DTEA, this value can be set to any of the above values except iLBC 20, or iLBC 30. For DTNA, this value should only be set to None, G711, A law, μ law, iLBC 20, or iLBC 30 otherwise an error message will be issued when the trunk is enabled.
Accept Inbound Transfer Requests
Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
Yes
No

Defaults
Yes

Explanation
This determines whether a request to transfer a call to a different SIP endpoint is to be accepted.
Add Host Name To User Agents?

Parameter group

VoIP SIP Signalling

Applicability

DTEA and DTNA

Access level

Admin

Possible values

No
Yes

Defaults

Yes

Explanation

If enabled, appends the hostname of the machine to the end of all User Agents used by registrations. The main purpose of this is to denote which machine an individual registration has come from.

User Agents are sent as specified by the following in order of priority, with the hostname appended to the result if enabled:

1. The row of the ini file that describes the destination.
2. The definition of Registrar in $SYS_DIR/voip/master.ini, if not specified in the row.
3. The entry in the System Configuration window, if neither of the above are specified.
Alarms - Make All Alertable

Parameter group
Application Server Interface

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
By default, some messages about alarm conditions are written to the AIX error log and others are written only in the WebSphere Voice Response errorlog. Select Yes if you want WebSphere Voice Response to write all alarm messages to the log, regardless of severity. If you select Yes, you must also set Alarms - Send to AIX Error Log to Yes.

Changes to this parameter do not take effect until you shut down WebSphere Voice Response, use the DT_shutdown command to stop dalarmd (the WebSphere Voice Response alarm daemon) and the SNMP daemon, then restart WebSphere Voice Response.

Note: This parameter was previously called NetView® - Send ALL Errors.
Alarms - Send to AIX Error Log
Parameter group
Application Server Interface

Access level
Admin

Possible values
No
Yes (EBCDIC codepoints)
Yes (ASCII codepoints)

Defaults
No

Explanation
Specifies whether WebSphere Voice Response logs alarm conditions in the AIX error log, to be passed on to NetView. (You do not need to set this parameter if you are using the SNMP traps that are generated by the WebSphere Voice Response SNMP daemon, dtsnmpd.)

Yes (EBCDIC codepoints)
Some or all WebSphere Voice Response alarm conditions are logged in the AIX error log with details in EBCDIC suitable to be passed by Alert Manager to an IBM program such as NetView/390.

Yes (ASCII codepoints)
Some or all WebSphere Voice Response alarm conditions are logged in the AIX error log with details in ASCII that can be sent to a workstation program such as NetView for AIX. For these alarm conditions to be notified to NetView for AIX, you must have trapgend installed.

You can use this parameter to cause WebSphere Voice Response alarm conditions to be logged in the AIX error log whether or not you are using NetView.

By default, only red alarm conditions and the corresponding clear messages are logged in the AIX error log. To log all alarm conditions, you must also set the Alarms - Make All Alertable parameter to Yes.
Changes to this parameter do not take effect until you shut down WebSphere Voice Response, use the `DT_shutdown` command to stop dtalarmd (the WebSphere Voice Response alarm daemon), then restart WebSphere Voice Response.

**Note:** In Version 1, this parameter was named NetView Installed.
Alert Level

Parameter group
ISDN Signaling

Access level
Field

Possible values
ISDN alerting
Call control alerting

Defaults
ISDN alerting

Explanation
This option allows you to select an alerting level for ISDN. When you select ISDN alerting, the alert is sent from layer 3 (fast). When you select call control, the alert is sent from layer 4.
Allow incoming numbers with Presentation Restricted

Parameter group

General

Access level

Field

Possible values

Disabled
Enabled

Defaults

Disabled

Explanation

When you select Enabled, applications can have access to calling party, originating numbers and redirected numbers when an Information Element (IE) is received with the Presentation attribute set to Presentation Restricted.
Answer Delay Time (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

20 through 5100

Defaults

260

Explanation

Specifies the minimum period WebSphere Voice Response delays before answering an incoming call. Answer Delay Time is used only with tie line protocols (E&M, U.K. TIE/DID, and R2 digital line signaling), and applies only if the Call Information Type parameter for the Channel Group is set to None or Signaling Process. It does not apply if Call Information Type is set to Register.

Some Central Office switches do not connect a voice path for some time after the call is answered. As a result, the first prompt played by WebSphere Voice Response might be cut off. To prevent this, increase the value so that the prompt is not cut off.
Answer Detect Threshold (dBm)

**Parameter group**

Trunk Interface

**Access level**

Admin

**Possible values**

-43 through -25

**Defaults**

-37

**Explanation**

Some CAS protocols do not provide positive answer supervision (for example, FXS, SAS, and Remote Extension (RE)). For this type of protocol, WebSphere Voice Response tries to detect voice energy to indicate that a call has been answered. Voice energy is characterized as being made up of more than two frequencies and has a different energy distribution from call progress tones (such as ring back or busy tones).

This parameter specifies the minimum level of audio signal (more than two frequencies) that must be present for a time equal to or greater than that specified by Answer Detect Time, to determine whether an outbound call from WebSphere Voice Response has been answered.

This parameter applies only to signaling protocols that do not return answer supervision.

**Note:** This parameter has been set to a default value that works well with a large variety of telephony installations. Reliable voice detection is a balance between rejecting noise or distorted call progress tones, and having enough sensitivity to detect a caller who speaks quietly or speaks only a short greeting.

If this value is set too low, you will get many false voice detections. If this value is set too high, the caller will have to speak quite loudly to cause voice detection.
**Answer Detect Time (ms)**

**Parameter group**

Trunk Interface

**Access level**

Admin

**Possible values**

Integer multiples of 20, in the range 0 through 10000

**Defaults**

60

**Explanation**

Some CAS protocols do not provide positive answer supervision (for example, FXS, SAS, and Remote Extension (RE)). For this type of protocol, WebSphere Voice Response tries to detect voice energy to indicate that a call has been answered. Voice energy is characterized as being made up of more than two frequencies and has a different energy distribution from call progress tones (such as ring back or busy tones). This parameter specifies the minimum amount of time that an audio signal (more than two frequencies) must be present at a level equal to or greater than that specified by Answer Detect Threshold (dBm), to determine whether an outbound call from WebSphere Voice Response has been answered.

This parameter applies only to signaling protocols that do not return answer supervision.

**Note:** This parameter has been set to a default value that works well with a large variety of telephony installations. Reliable voice detection is a balance between rejecting noise or distorted call progress tones and having enough sensitivity to detect a caller who speaks quietly or speaks only a short greeting.

If this value is set too short, you will get many false voice detections. If this value is set too long, the caller will have to speak for a long time to cause voice detection.
Area Code

Parameter group

Channel Group

Access level

Admin

Possible values

A 1- to 6-character string (0 through 9, A, B, C, and D) or blank

Defaults

408

Explanation

Specifies the area code prefix for all channels in this group. The area code can be used to start the correct voice application in response to an incoming telephone call. (For more information about how area code is used, see “Channel identification” on page 37)

The way WebSphere Voice Response uses the value of Area Code depends on the value of the Call Information Type parameter for the Channel Group (see “Call Information Type” on page 225 for details).

This value can be any number that you select to identify with all the channels in a channel group, up to six characters long. The valid characters are the digits 0 through 9 and the letters A, B, C, and D.
Audio Name CompressionType  
Parameter group

Application Server Interface

Access level

Admin

Possible values

Compressed
Uncompressed

Defaults

Compressed

Explanation

Specifies whether audio names are recorded compressed or uncompressed, when using the RecordAudioName state table action.

Set this parameter when you install WebSphere Voice Response, and do not to change it afterward. This precaution prevents you having some audio names that are compressed and some that are uncompressed on the same system. It also helps WebSphere Voice Response to search more efficiently.

An uncompressed audio name occupies five times more disk space than the same audio name does when compressed.

If much interference exists when the audio name is recorded, you might find that compressing the audio name results in a poor quality of playback.

If you want to change this parameter from its default setting, ensure that all your custom servers that use the CA_Get_Audio_Name_Info subroutine can process a returned AUDIO_NAME_INFO_ST structure in which the compression_type field can have any valid value. If you do not, the data returned will be corrupted.

Note: Do not change this parameter from its default setting if you are using the DirectTalkMail feature of DirectTalk for AIX Version 2.1.

Restart your WebSphere Voice Response system after changing this parameter.
Backup Time and Erase after DTMF (Interrupts)

Parameter group

Trunk Interface

Access level

Admin

Possible values

0 through 8 units

Defaults

5 units (100 ms)

Explanation

Specifies how much is erased at the end of a recorded voice segment to remove any noise that was caused when the key that signals the end of recording was pressed. For example, when you record a message or a voice segment, and press * to signal that you have completed, * might introduce noise into the message or voice segment. WebSphere Voice Response then “rewinds” for the amount of time specified by this parameter so that the noise is not part of the voice segment and the next recording automatically erases it.

This value is adjustable in units of 20 ms. The default value sets this parameter to 100 ms.
B-Channel Service Message Support

Parameter group

ISDN signaling

Access level

Admin

Possible values

Yes
No

Defaults

No

Explanation

Selecting yes will enable B-channel service messages to be supported for certain ISDN protocols and switches. B-channel service messages allow individual channels to be taken in and out of service.

This will affect Nortel DMS 10, DMS 250 and Avaya 5E12 switches.

Note: TR41449/TR41459 always use B-channel service messages regardless of this parameter.
Blocking Action
Parameter group

Signaling Type

Access level
Field

Possible values
Offhook
Other

Defaults
Offhook

Explanation
Specifies how WebSphere Voice Response indicates that the Operating Status of a channel is Blocked. This is used to prevent a channel from receiving incoming calls. Select Offhook if your method of blocking a channel is to make it seem busy to the switch. Select Other if:

- Your method of blocking a channel is not to respond with a wink start or delay start signal to an incoming seizure from the switch (used with Aspect CallCenter E&M signaling).

or:
- The T1 CAS Protocol is set to FXS and the FXS Start Type parameter is set to Ground Start (in this condition, the blocking action is a continuous ground on the ring lead).

Note that Other applies only when the T1 CAS Protocol parameter is set to one of the following:
- DID (and DID Start Type is set to Wink Start or Delay Start)
- E&M or ROLM E&M (and E&M Start Type is set to Wink Start or Delay Start)
- FXS (and FXS Start Type is set to Ground Start: for Loop Start, select Offhook.)
Buffer Pool Address

Parameter group

General

Access level

Admin

Possible values

0x30000000 through 0xA0000000

Defaults

0x40000000

Explanation

Specifies the address at which WebSphere Voice Response attaches its buffer pool shared memory segment. If you have another software package used in a custom server and that attaches shared memory at a fixed location that is also used by WebSphere Voice Response, you can use this parameter to specify a different location for WebSphere Voice Response to attach its segment.

The value must not be the same as the value of either Control Memory Address or Error Table Address.
Cadence Energy Maximum (dBm)

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

-36 through 0

**Defaults**

0

**Explanation**

Specifies, together with Cadence Energy Minimum, the energy band inside which a cadenced hang-up is to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
**Cadence Energy Minimum (dBm)**

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

-36 through 0

**Defaults**

-28

**Explanation**

Specifies, together with Cadence Energy Maximum, the energy band inside which a cadenced hang-up is to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
Cadence Off Time Maximum (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

Integer multiples of 20, in the range 80 through 10000

Defaults

600

Explanation

Specifies the maximum length of silence that must occur between two signals for a cadenced hang-up to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
Cadence Off Time Minimum (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

Integer multiples of 20, in the range 80 through 10000

Defaults

400

Explanation

Specifies the minimum length of silence that must occur between two signals for a cadenced hang-up to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
**Cadence On Time Maximum (ms)**

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

Integer multiples of 20, in the range 80 through 10000

**Defaults**

600

**Explanation**

Specifies the maximum length of signal (in the energy band specified by Cadence Energy Minimum and Cadence Energy Maximum) that must occur for a cadenced hang-up to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
**Cadence On Time Minimum (ms)**

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

Integer multiples of 20, in the range 80 through 10000

**Defaults**

400

**Explanation**

Specifies the minimum length of signal (in the energy band specified by Cadence Energy Minimum and Cadence Energy Maximum) that must occur for a cadenced hang-up to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
Cadence Silence Maximum (dBm)

Parameter group
Signaling Type

Access level
Admin

Possible values
-36 through 0

Defaults
-28

Explanation
Specifies the maximum energy level to be interpreted as the silence that must occur between two signals for a cadenced hang-up to be detected. For more information, see “Setting parameters for hangup tone detection” on page 112.
Call Detail Record Logging
Parameter group

General

Access level
Admin

Possible values
No
cdr.log
DB2
Both

Defaults
No

Explanation
Specifies where WebSphere Voice Response is to log call detail records.
No Call detail records are not written.
cdr.log Call detail records are written to a log file called $CUR_DIR/oamlog/cdr.log but not to the DB2 database.
DB2 Call detail records are written to the DB2 database, but not to $CUR_DIR/oamlog/cdr.log.
Both Call detail records are written to a log file called $CUR_DIR/oamlog/cdr.log and to the DB2 database.
Call Information Type

Parameter group

Channel Group

Access level

Admin

Possible values

None
Signaling Process
Register

Defaults

None

Explanation

Specifies the source of call information. Select Register when the called number and/or the calling number are provided via the signaling channel (dial pulse) or voice channel (DTMF or MFR1). Note that Register is used only with channel associated signaling (CAS) protocols.

Select Signaling Process when call information is provided on the exchange data link, or by a common channel signaling (CCS) protocol such as ISDN, or some other custom-written signaling process. When you select Signaling Process, ensure that you select the correct value for the Signaling Process Type parameter (in the Channel Group parameter group).

Select None when no call information is provided.

The value of call information type determines how incoming calls are identified, as follows:

Register
WebSphere Voice Response uses only the received called number (DID/DNIS) to identify the correct voice application; it does not use the value of the Area Code parameter.

Signaling Process
WebSphere Voice Response uses the value of the Area Code parameter and the called number provided by the signaling process to identify the correct voice application.
None  The called number is not expected. WebSphere Voice Response uses the value of the Area Code parameter and the value of the Phone Number parameter (in the Channel parameter group) to identify each channel and distinguish calls in relationship to the channels on which they arrive.

When Call Information Type is set to **Signaling Process** or **Register** but, on a given call, the called number is not received, WebSphere Voice Response uses the value of the Area Code parameter and the value of the Phone Number parameter (in the Channel parameter group) to identify each channel and distinguish calls in relationship to the channels on which they arrive. This would otherwise be an error condition.

When Call Information Type is set to **Signaling Process** or **None** and the E1 CAS Protocol parameter is set to E&M/US or UK Tie/DDI, set the Start Type parameter to Immediate Start because no address signaling is sent in either the signaling channel or in-band via DTMF.
Call Signaling Port

Parameter group

VoIP SIP Signaling

Access level

Admin

Possible values

1024 through 65535

Defaults

5060

Explanation

The local port used for all SIP signaling.
Called Number Character to Strip
Parameter group

Exchange Data Link

Access level
Admin

Possible values
Alphanumeric character

Defaults
0 (ASCII zero)

Explanation
Specifies the character used by the switch to pad the called number to form a fixed-length string containing the number. This parameter is used only if the Called Number Stripping parameter is set to Yes.
Called Number Length
Parameter group
Exchange Data Link

Access level
Admin

Possible values
1 through 64

Defaults
7

Explanation
Specifies the number of digits in the called number that WebSphere Voice Response receives over the exchange data link.
**Called Number Length (Minimum)**

**Parameter group**
Exchange Data Link

**Access level**
Admin

**Possible values**
1 through 64

**Defaults**
7

**Explanation**
Specifications the minimum number of digits in the called number after any padding characters have been stripped. This parameter is used only if the Called Number Stripping parameter is set to Yes.
**Called Number Stripping**

**Parameter group**

Exchange Data Link

**Access level**

Admin

**Possible values**

Yes

No

**Defaults**

No

**Explanation**

Specifies whether WebSphere Voice Response expects the called number to be padded. If you select No, WebSphere Voice Response expects the numbers not to be padded and handles each one as the actual called number.

If you select Yes, WebSphere Voice Response strips from the called number any leading characters that match the character that is set by Called Number Character to Strip. This continues until the number reaches the length defined by Called Number Length (Minimum) or a character in the number does not match the Called Number Character to Strip. If the called number does not contain a leading character that matches the character that is set by Called Number Character to Strip, WebSphere Voice Response assumes the called number is not padded and handles it as such.

The Called Number Stripping parameter allows variable-length numbers to be received within an exchange data link protocol that supports only fixed length-numbers.
Calling Number Character to Strip
Parameter group
Exchange Data Link

Access level
Admin

Possible values
Alphanumeric character

Defaults
0 (ASCII zero)

Explanation
Specifies the character that is used by the switch to pad the calling number to make a fixed-length string containing the number. This parameter is used only if the Calling Number Stripping parameter is set to Yes.
Calling Number Length
Parameter group

Exchange Data Link

Access level
Admin

Possible values
1 through 64

Defaults
7

Explanation
Specifies the number of digits in the calling number information that WebSphere Voice Response receives over the exchange data link.
**Calling Number Length (Minimum)**

**Parameter group**

Exchange Data Link

**Access level**

Admin

**Possible values**

1 through 64

**Defaults**

7

**Explanation**

Specifies the minimum number of digits in the calling number after any padding characters have been stripped. This parameter is used only if the Calling Number Stripping parameter is set to Yes.
Calling Number Stripping
Parameter group

Exchange Data Link

Access level
Admin

Possible values
Yes
No

Defaults
No

Explanation
Specifies whether WebSphere Voice Response expects the calling number to be padded. If you select No, WebSphere Voice Response expects the numbers not to be padded and handles each one as the actual calling number.

If you select Yes, WebSphere Voice Response strips from the calling number any leading characters that match the character that is set by Calling Number Character to Strip. This action continues until the number reaches the length defined by Calling Number Length (Minimum), or a character in the number does not match the Calling Number Character to Strip. If the calling number does not contain a leading character that matches the character that is set by Calling Number Character to Strip, WebSphere Voice Response assumes the calling number is not padded and handles it as such.

The Calling Number Stripping parameter allows variable-length numbers to be received in an exchange data link protocol that supports only fixed-length numbers.
Calling Party Number — MWI Identification

Parameter group

ISDN signaling

Access level

Admin

Possible values

A text string containing up to 40 characters. Valid characters are 0 through 9, A, B, C, D, *, and #.

Defaults

No

Explanation

If this calling party number is set, it will appear in an outgoing ISDN call setup message. The calling party number is used to identify which WebSphere Voice Response client is requesting to set or clear a Message Waiting indicator.

In this case, the MWI is being controlled over the D-channel and the feature is currently available only for the ISDN QSIG signal process.
CAS - Allow Alternate Hangup
Parameter group

Channel Group

Access level
Admin

Possible values
No  Yes

Defaults
No

Explanation

Some channel associated signaling (CAS) protocols (such as E&M and R2) provide positive far-end hang-up detection by using the signaling bits. This parameter specifies whether WebSphere Voice Response is to take notice of other sources of far-end hang-up indication, such as constant or cadenced energy tone detection or exchange data link signaling. This parameter is ignored if a common channel signaling (CCS) protocol is being used, or if the signaling protocol does not provide positive far-end hang-up detection.

No  If the signaling protocol supports far-end hang-up detection using the signaling bits, WebSphere Voice Response ignores other sources of far-end hang-up indication. For example, even if a constant hang-up tone is detected, a state table continues.

Yes  WebSphere Voice Response takes notice of other sources of far-end hang-up indication whether or not the signaling protocol supports far-end hang-up detection using the signaling bits. For example, when a constant hang-up tone is detected, a state table ends (assuming that the Hang Up Detection parameter is set to Constant Energy Detection).
CCS Clustered mbufs in Receive Pool

Parameter group

Trunk Interface

Access level

Admin

Possible values

5 through 200

Defaults

10

Explanation

Specifies the number of clustered kernel memory buffers (mbufs) to be preallocated to a pool of buffers when a signaling channel is opened on a WebSphere Voice Response PCI adapter (such as a DTTA.) This mbuf pool is used to hold large incoming common channel signaling (CCS) packets until the signaling process is ready to read them. Another buffer pool is available to hold small data packets. Each unit represents one clustered kernel memory buffer (256 bytes of kernel memory, with a 4 Kbyte page attached).
CCS mbufs in Receive Pool

Parameter group

Trunk Interface

Access level

Admin

Possible values

40 through 2000

Defaults

128

Explanation

Specifies the number of kernel memory buffers (mbufs) to be preallocated to a pool of buffers when a signaling channel is opened on a WebSphere Voice Response PCI adapter (such as a DTTA.) This mbuf pool is used to hold small incoming common channel signaling (CCS) packets until the signaling process is ready to read them. Another buffer pool is available to hold large data packets. Each unit represents one kernel memory buffer (256 bytes of pinned kernel memory).
CCS Signaling Link Mode
Parameter group
Trunk Interface

Access level
Field

Possible values
Backup
None
Primary

Defaults
None

Explanation
Specifies, when a common channel signaling (CCS) protocol is being used, whether the trunk is to be used as the primary signaling channel, backup signaling channel (D-channel backup), or neither. Select **Primary** if this trunk is the primary signaling channel. Select **Backup** if this trunk is to be the backup channel. (If the protocol is ISDN, this applies only if D-channel backup is in use.) Select **None** if ISDN non-facility associated signaling (NFAS) is in use and this trunk is not to carry signaling information. This parameter does not apply to channel associated signaling (CAS).
Channel Group
Parameter group

Channel

Access level
Admin

Possible values
1 through 16

Defaults
1

Explanation
Specifies the channel group to which this channel belongs.
Check Voice Messages Time - Alert (ms)

Parameter group
Application Server Interface

Access level
Admin

Possible values
10 through 20000 milliseconds

Defaults
1500 milliseconds

Explanation
Every time the time taken to retrieve voice messages exceeds the value of this parameter, WebSphere Voice Response logs white (information) message 1403.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Check Voice Messages Time - Max Allowable (ms)

Parameter group
Application Server Interface

Access level
Admin

Possible values
10 through 20000 milliseconds

Defaults
1000 milliseconds

Explanation
Specifies a maximum average time to retrieve voice messages. If the average time exceeds this value, a white (information) message 25206 is logged.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Check Voice Messages Time - Recovered (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

10 through 20000 milliseconds

Defaults

800 milliseconds

Explanation

Specifies the average retrieval time for voice messages to which the system must return before WebSphere Voice Response generates green (cleared) message 25207. This follows a condition where the system has exceeded the maximum allowable retrieval time and logged message 25206.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see "CHP Performance Metrics - Expiry Time (mins)" on page 247) is set to a value greater than 0.

For more information, see "Monitoring the performance of a single system image" on page 150.
**CHP available call reject threshold**  
**Parameter group**  
VoIP SIP Signalling  

**Applicability**  
DTEA and DTNA  

**Access level**  
Admin  

**Possible values**  
0  
A integer greater than 0  

**Default**  
0  

**Explanation**  
To allow sufficient time for handling new incoming calls while other CHPs are cleaning up from previous calls, this parameter determines the number of additional WebSphere Voice Response Channel Processes (CHPs) that must be available before an incoming call can be accepted. If the arrival rate of new incoming calls exceeds the rate at which CHPs are being cleaned up, SIP calls may not get answered within the required time causing timeouts and recovery actions to be instigated.

If set to zero (default), no check is made as to whether at least one CHP is available before accepting the SIP Call. If set to any other value, the number of free (unallocated) CHPs must be more than that value for the SIP call to be accepted.

If a call is rejected for this reason, a '486 Busy' is returned as a response to the SIP incoming call. This causes the far end to retry rather than timeout, cancel and restart the call.
CHPM Socket Port Number

Parameter group
Application Server Interface

Access level
Admin

Possible values
0 through 65535

Defaults
26923

Explanation
Specifies the UDP port number at which CHPM listens for requests from WebSphere Voice Response for Java. You need to change this value only if the default port number is in use by another program. If you change this value and you have WebSphere Voice Response for Java running on this system, you must also change the AIXPortNumber configuration parameter for this host in the WebSphere Voice Response for Java configuration to match it.
CHP Performance Metrics - Expiry Time (mins)

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 60 minutes

Defaults

3 minutes

Explanation

Controls how WebSphere Voice Response collects measurements on CHP performance. These measurements include play latency time, underrun margin time, voice message retrieval time, and profile retrieval time.

Set this parameter as follows:

0 Data is not collected.

>0 Data is collected. However, when WebSphere Voice Response calculates any averages of measurements taken across many channel processes, it ignores any measurements that are older than the value of this system parameter.

For more information, see "Monitoring the performance of a single system image" on page 150.
CHP Performance Metrics - Weighting of Old Average

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 9999

Defaults

15

Explanation

Specifies the weighting that WebSphere Voice Response gives to the existing average when it takes a new measurement and calculates a new average. This applies to all the CHP performance measurements: play latency time, underrun margin time, voice message retrieval time, and profile retrieval time.

If you specify a low value for this parameter, new measurements affect the average quickly. For example, the value 0 makes the average equal to the existing measurement, and the value 1 takes the average of the last 2 measurements.

If you specify a higher value for this parameter, more measurements are included when WebSphere Voice Response calculates the average.

For more information, see “Monitoring the performance of a single system image” on page 150.
CO Acknowledgment (ms)

Parameter group

Signaling Type

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 5100

Defaults

60

Explanation

Specifies the minimum duration for a signal from the switch to determine whether the signal is valid. This parameter is used to qualify all signals coming from the switch except off-hook and on-hook signals. It is used for incoming and outgoing calls. Note that the parameter does not apply to the EL7/CAS (Ericsson MD110) signaling protocol.
## CO Off Hook (ms)

### Parameter group
- Signaling Type

### Access level
- Field

### Possible values
Integer multiples of 20, in the range 0 through 5100

### Defaults
- 60

### Explanation
Specifies the minimum duration for an off-hook signal from the switch to determine whether the signal is valid. It is used for incoming and outgoing calls. Note that the parameter does not apply to the EL7/CAS (Ericsson MD110) signaling protocol.
CO On Hook (ms)

Parameter group

Signaling Type

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 5100

Defaults

200

Explanation

Specifies the minimum duration for an on-hook signal from the switch to determine whether the signal is valid. It is used for incoming and outgoing calls. Note that the parameter does not apply to the EL7/CAS (Ericsson MD110) signaling protocol.

When the T1 CAS Protocol parameter is set to SAS or FXS (and the FXS Start Type parameter is set to Loop Start), the value of CO On Hook also affects recognition of a disconnect clear (wink off) signal. A disconnect clear signal, if provided by a switch, is a momentary change of state of the a-bit. WebSphere Voice Response interprets the change of state as a far-end disconnect, if the change lasts for the period specified by the CO On Hook parameter.
Connect Voice Channel Before Answer

Parameter group

Channel Group

Access level

Field

Possible values

Disabled
Enabled

Defaults

Disabled

Explanation

This parameter applies to incoming calls and governs whether voice can be played by WebSphere Voice Response before the call has been answered. This allows announcements to be played to a caller before charging is applied. To avoid breaking homologation rules, this parameter should only be enabled for channels of channel groups configured to receive calls inside a telephone company's network. These would typically be configured to be handled by an SS7 protocol.
Constant Energy Maximum (dBm)

Parameter group

Signaling Type

Access level

Admin

Possible values

-36 through 0

Defaults

0

Explanation

Specifies, with Constant Energy Minimum, the energy band in which a constant tone is detected as hang-up. The energy must be constant to inside 1 dBm to be recognized as hang-up.
Constant Energy Minimum (dBm)

Parameter group

Signaling Type

Access level

Admin

Possible values

-36 through 0

Defaults

-24

Explanation

Specifies, with Constant Energy Maximum, the energy band within which a constant tone is detected as hang-up. The energy must be constant to inside 1 dBm to be recognized as hang-up.
Control Memory Address
Parameter group

General

Access level
Admin

Possible values
0x30000000 through 0xA0000000

Defaults
0x30000000

Explanation
Specifies the address at which WebSphere Voice Response attaches its control shared memory segment. If you have another software package used in a custom server, which attaches shared memory at a fixed location that conflicts with the locations used by WebSphere Voice Response, you can use this parameter to specify a different location for WebSphere Voice Response to attach its segment.

The value must not be the same as the value of either Buffer Pool Address or Error Table Address.
Country or Region Parameter group

General

Access level

Field

Possible values

- Argentina
- Australia
- Austria
- Belgium
- Brazil
- Chile
- China
- Columbia
- Denmark
- Finland
- France
- Germany
- China (Hong Kong S.A.R.)
- Italy
- Japan
- Monaco
- New Zealand
- Norway
- Paraguay
- Peru
- Portugal
- Singapore
- South Africa
- Spain
- Sweden
- Switzerland
- The Netherlands
- US/Canada
- United Kingdom
- Unassigned
- Venezuela

Defaults

Unassigned

Explanation

The country or region in which the WebSphere Voice Response system is installed. This parameter is used only by Pack Configuration to determine the values of other system parameters.

Note: The absence of a country or region in this list does not mean that WebSphere Voice Response cannot operate there but, the presence of a country or region does not necessarily mean that WebSphere Voice Response is approved by the local telecommunications authority for use. Contact your IBM representative for current information.
CPU Clear

Parameter group

CPU Monitor

Access level

Admin

Possible values

0 through 100 percent

Defaults

50 percent

Explanation

When the monitor detects CPU idle time that exceeds this percentage, it sends a message to the console to clear a previous CPU warning or alert.
CPU Warning Threshold
Parameter group

CPU Monitor

Access level
Admin

Possible values
0 through 100 percent

Defaults
5 percent

Explanation
When the monitor detects that CPU idle time has fallen to this percentage, it displays a yellow alert on the System Monitor Console to warn that the system is running almost at full capacity.
Database Availability Check Timeout

Parameter group
Application Server Interface

Access level
Admin

Possible values
3 through 60 seconds

Defaults
15 seconds

Explanation
If DB2 does not respond to a query correctly inside this time a red error is logged and the defined action is taken for phone calls. If all is well, another check is made after a quarter of this time with a minimum interval of one second. After an error, the interval is one second.
DBIM Time Out

Parameter group

General

Access level

Admin

Possible values

5 through 600

Defaults

20

Explanation

Specifies, in seconds, how long a process is to wait for a response from the voice database. When the specified amount of time has elapsed, the request times out, and is lost.
D-Channel Service Message Support
Parameter group

ISDN signaling

Access level

Admin

Possible values

Yes
No

Defaults

No

Explanation

Selecting yes will enable D-channel Service Messages to be supported by the ISDN system. D-channel service messages are required for D-channel backup. D-channel service messages are used to communicate changes of state of the signaling channel to in service, out of service and standby states.

Note: If backup trunks are configured, this parameter will be automatically enabled.
Default **CLID for Incoming VoIP calls**

**Parameter group**

VoIP SIP Signaling

**Access level**

Admin

**Possible values**

Numeric String

**Defaults**

6661234

**Explanation**

The default Calling Line ID (CLID) to use for application profile selection when a suitable identifier cannot be found in an incoming call.
Default Destination URI
Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
A character string.

Defaults
Null

Explanation
The routing address for messages when the Proxy Mode parameter is set to None. This may be the address of a SIP gateway. If this field is left blank, calls are sent directly to the endpoint specified in the To Header.

Example formats include:
• sip:gateway@anyplace.com
• sip:gateway@9.20.38.97
Default Destination Port
Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
1024 through 65535

Defaults
5060

Explanation
The port associated with the default destination URI.
Default Diskette Drive
Parameter group

General

Access level
Admin

Possible values
A device name

Defaults
/dev/rfd0

Explanation
Specifies the diskette drive that WebSphere Voice Response utilities (such as import and export) use by default.
Default RTP router
Parameter group
VoIP Media - Adapters

Access level
Admin

Possible values
0.0.0.0 through 255.255.255.255

Defaults
0.0.0.0

Explanation
Records the IPv4 IP address of the default RTP router for a DTEA card. There are four instances of this parameter, one for each card.
Default System Prompt Directory Name

Parameter group

Application Server Interface

Access level

Admin

Possible values

A prompt directory name

Defaults

System

Explanation

Specifies the prompt directory that is used as the default by voice applications. The default value identifies the directory that is delivered with WebSphere Voice Response. If you decide to designate a different directory as the default directory, or if you rename System, you must reset this parameter.
**Default Tape Drive**

**Parameter group**

WebSphere Voice Response

**Access level**

Admin

**Possible values**

A device name

**Defaults**

`/dev/rmt0`

**Explanation**

Specifies the tape drive that WebSphere Voice Response utilities (such as import and export) use by default.
Delay Start Delay (ms)
Parameter group

Signaling Type

Access level
Field

Possible values
20 through 140

Defaults
120

Explanation
Specifies how much time elapses after WebSphere Voice Response receives the off-hook signal from the switch but before it sends a delay start signal to the switch. The delay start delay is only important when channels are using delay start as the start type. The start type is determined by the correct Start Type parameter in the Channel Group parameter group.
Delay Start Duration (ms)
Parameter group

Signaling Type

Access level
Field

Possible values
140 through 4000

Defaults
200

Explanation
Specifies the length of the delay start signal that is sent to the switch. The length of the delay start signal is important only when channels are using delay start as the start type. The start type is determined by the correct Start Type parameter in the Channel Group parameter group.
Dial Pause (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

20 through 10000

Defaults

200

Explanation

Specifies how long WebSphere Voice Response waits before taking action if the Dial Tone Detection parameter (Channel Group) is set to No (dial pause).

Note: Dial tone detection is enabled for loop start channels, whatever the setting of the Dial Tone Detection parameter.
Dial Tone Detection

Parameter group

Channel Group

Access level

Admin

Possible values

No (dial pause)
Yes (dial tone)

Defaults

No (dial pause)

Explanation

Specifies whether WebSphere Voice Response is to expect dial tone from the switch. If this value is set to No (dial pause), WebSphere Voice Response does not expect dial tone from the switch. WebSphere Voice Response takes action, for example, by sending outgoing address signals after the time specified for the Dial Pause parameter in the Signaling Type parameter group.

If this value is set to Yes (dial tone), WebSphere Voice Response does not take action until a valid dial tone is received from the switch. Dial tone, as defined by the correct call progress tone parameters, must be present for the amount of time specified by the Dial Tone Qualify Time parameter in the Signaling Type parameter group.

Note that dial tone detection is enabled for loop start channels, whether or not this parameter is set to No (dial pause).
Dial Tone Qualify Time (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

0 through 5100

Defaults

200

Explanation

Specifies how long WebSphere Voice Response listens to a call progress tone from the switch to qualify it as a valid dial tone. This parameter applies only when Dial Tone Detection (Channel Group) is set to Yes (dial tone).

Note: This parameter was called Dial Tone in previous releases.
Dial Tone Timeout (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

0 through 60000

Defaults

8000

Explanation

Specifies how long WebSphere Voice Response waits for dial tone before assuming that a problem has occurred. This parameter applies only when Dial Tone Detection parameter for the Channel Group is set to Yes (dial tone).
DID Start Type

Parameter group

Channel Group

Access level

Admin

Possible values

Immediate Start
Delay Start
Wink Start

Defaults

Wink Start

Explanation

Specifies the start signal that WebSphere Voice Response uses for all channels in the channel group, for incoming address signaling.

With Wink Start or Delay Start operation, address digits are not sent until after the start signal (off-hook/on-hook) is received by the switch. Immediate Start allows address signals to be sent immediately (for example, inside 65 ms) after seizure by the switch.

If the Call Information Type parameter for the Channel Group is set to Register, only wink start or delay start operation are recommended. With immediate start, a possibility exists that address digits might be missed if a register is not attached before digits are sent. However, immediate start operation is best if Call Information Type is set to Signaling Process or None.
Direction

Parameter group

Channel Group

Access level

Admin

Possible values

Incoming
Outgoing
Bothway

Defaults

Bothway

Explanation

Specifies whether channels receive calls only, make calls only, or do both. To debug a state table, at least one channel in the system must be configured to make calls (either Outgoing or Bothway).
DNSSRV Server Address

Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
A character string

Defaults
Null

Explanation
The address of the DNSSRV server, which is used when the Proxy Mode parameter is set to Automatic Routing: DNSSRV. Example formats include: dnssrv.ibm.com 9.20.38.97
DNSSRV Server Port

Parameter group

VoIP SIP Signaling

Access level

Admin

Possible values

1024 through 65535

Defaults

5061

Explanation

The IP port of the DNSSRV server.
**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

20 through 140

**Defaults**

100

**Explanation**

Specifies the maximum amount of break time required for WebSphere Voice Response to detect a received dial pulse. Whether the signaling pattern is recognized and interpreted as a dial pulse is determined by the value of this parameter in combination with the values of DP Receive Minimum Break, DP Receive Minimum Make, and DP Receive Maximum Make.
DP Receive Maximum Make (ms)  
Parameter group  
Trunk Interface  
Access level  
Field  
Possible values  
20 through 100  
Defaults  
60  
Explanation  
Specifies the maximum amount of make time required for WebSphere Voice Response to detect a received dial pulse. Whether the signaling pattern is recognized and interpreted as a dial pulse is determined by the value of this parameter and the values of DP Receive Minimum Make, DP Receive Minimum Break, and DP Receive Maximum Break.
DP Receive Minimum Break (ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

20 through 100

Defaults

20

Explanation

Specifies the minimum amount of break time required for WebSphere Voice Response to detect a received dial pulse. Whether the signaling pattern is recognized and interpreted as a dial pulse is determined by the value of this parameter and the values of DP Receive Maximum Break, DP Receive Minimum Make, and DP Receive Maximum Make.
DP Receive Minimum Make (ms)

Parameter group
Trunk Interface

Access level
Field

Possible values
20 through 100

Defaults
20

Explanation
Specifies the minimum amount of make time required for WebSphere Voice Response to detect a received dial pulse. Whether the signaling pattern is recognized and interpreted as a dial pulse is determined by the value of this parameter and the values of DP Receive Maximum Make, DP Receive Minimum Break, and DP Receive Maximum Break.
DP Transmit Break (ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

20 through 100

Defaults

60

Explanation

Specifies the break (on-hook) time for dial pulses transmitted by WebSphere Voice Response to the switch.
DP Transmit Speed (pulse/sec)
Parameter group
Trunk Interface

Access level
Field

Possible values
4 through 25

Defaults
10

Explanation
Specifies the speed at which dial pulses are sent for address signaling for outgoing calls that are made by using dial pulsing (rotary or decadic dialing). The unit of measure is pulses per second.
DTMF Algorithm Variant
Parameter group

Trunk Interface

Access level

Field

Possible values

Normal Mode
DTMF Algorithm Variant 1

Defaults

Normal Mode

Explanation

This parameter controls which algorithm is used for DTMF detection. DTMF Algorithm Variant 1 is a slight change to the Normal DTMF detection algorithms used on the DTTA to reduce spurious detection of DTMF in the presence of speech and music. However, using DTMF Algorithm Variant 1 may result in some previously detected DTMFs that were close to the edge of the specification for DTMFs being rejected.
DTMF Maximum Receive Level (dBm)

Parameter group

Trunk Interface

Access level

Admin

Possible values

-7 through 0

Defaults

0

Explanation

Specifies the upper detection threshold for DTMF tones received by WebSphere Voice Response except when a Record... action such as RecordVoiceMessage is active.
DTMF Minimum Receive Level (dBm)

Parameter group

Trunk Interface

Access level

Admin

Possible values

-43 through -10

Defaults

-43

Explanation

Specifies the lower detection threshold for DTMF tones that are received by WebSphere Voice Response except when a Record... action such as RecordVoiceMessage is active.
DTMF Transmission Method

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA only

Access level

Admin

Possible values

In-band
RTP payload

Defaults

In-band

Explanation

Determines which method is used to send DTMF keys. This can be used when sending compressed voice to ensure that DTMF keys can be detected.
DTMF Transmit Level, Low Frequency (dBm)

Parameter group

Trunk Interface

Access level

Field

Possible values

-15 through -3

Defaults

-8

Explanation

Specifies the level at which WebSphere Voice Response transmits tones in the low-frequency DTMF group.
DTMF Transmit Level Twist (dBm)

Parameter group

Trunk Interface

Access level

Field

Possible values

0 through 5

Defaults

2

Explanation

Specifies the difference between the level at which WebSphere Voice Response transmits tones in the low frequency DTMF group (specified by DTMF Transmit Level, Low Frequency) and the transmit level for tones in the high-frequency DTMF group.

High-frequency group tones are always transmitted at a level greater than low-frequency group tones are. For example, with default values for DTMF Transmit Level, Low Frequency (-8 dBm) and DTMF Transmit Level Twist (2 dBm), the transmit level of tones in the DTMF high-frequency group is -8 + 2 = -6 dBm.
**DTMF Transmit On (ms)**

**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

40 through 220

**Defaults**

50

**Explanation**

Specifies how long a DTMF tone that WebSphere Voice Response transmits to the switch stays on.
DTMF Transmit Speed (digits/sec)

Parameter group

Trunk Interface

Access level

Field

Possible values

4 through 12

Defaults

10

Explanation

Specifies the speed at which DTMF digits are transmitted for outgoing address signaling. The unit of measure is digits per second.
DTTA Interrupt Separation Clear Threshold (ms)

Parameter group

Application Server Interface

Access level

Field

Possible values

1 through 20

Defaults

4

Explanation

Specifies the interval that should elapse between interrupts from two or more DTTAs for a green alarm (27079) to be generated. This green alarm clears the previous yellow alarm (27078) that was raised because the interval between interrupts fell below the value specified by the DTTA Interrupt Separation Warning Threshold parameter.

Do not reset this parameter unless requested by IBM Support.
DTTA Interrupt Separation Warning Threshold (ms)

Parameter group

Application Server Interface

Access level

Field

Possible values

1 through 20

Defaults

3

Explanation

Specifies the minimum acceptable interval that should elapse between interrupts from two or more DTTAs. If interrupts are too close together, a yellow alarm (27078) is generated. The yellow alarm is cleared when the interval exceeds the value specified by the DTTA Interrupt Separation Clear Threshold parameter.

Do not reset this parameter unless requested by IBM Support.
DTTA Loading Clear Threshold (%)
Parameter group
Application Server Interface

Access level
Field

Possible values
0 through 120

Defaults
90

Explanation
Specifies the loading on the DTTA for a green alarm (27075) to be generated. This green alarm clears the previous yellow alarm (27074) that was raised because the loading exceeded the value specified by the DTTA Loading Warning Threshold parameter.

Do not reset this parameter unless requested by IBM Support.
DTTA Loading Warning Threshold (%)

Parameter group

Application Server Interface

**Access level**

Field

**Possible values**

0 through 120

**Defaults**

95

**Explanation**

Specifies the maximum loading on the DTTA for a yellow alarm (27074) to be generated. The yellow alarm is cleared when the loading falls below the value specified by the DTTA Loading Clear Threshold parameter.

Do not reset this parameter unless requested by IBM Support.
E&M Start Type

Parameter group

Channel Group

Access level

Admin

Possible values

Immediate Start
Delay Start
Wink Start

Defaults

Wink Start

Explanation

Specifies the start signal that WebSphere Voice Response uses for all channels in the channel group, for both incoming and outgoing address signaling.

With wink start or delay start operation, address digits are not sent until after the start signal (off-hook/on-hook) is received by the switch. Immediate start allows address signals to be sent immediately (for example, inside 65 ms) after seizure by the switch.

If the Call Information Type parameter for the Channel Group is set to Register, only wink start or delay start operation are recommended. With immediate start, a possibility exists that address digits might be missed if a register is not attached before digits are sent. However, immediate start operation is best if Call Information Type set to values other than Register.
E1 CAS Protocol

Parameter group

Channel Group

Access level

Admin

Possible values

E&M/US
EL7/CAS (Ericsson MD110)
FXS
Italy
RE
R2
SL
TS003
UK Tie/DDI
UK Callstream
UK Exchange

Defaults

R2

Explanation

Specifies the E1 channel associated signaling (CAS) protocol that is used by channels in this group. This parameter is not used with common channel signaling (CCS).

Note: UK Tie/DDI can also be used whenever “inverted” E&M/US (also known as European E&M) signaling is required. Because it is not possible to select Wink Start for the UK Tie/DDI Start Type parameter when required, wink start operation can be simulated by setting UK Tie/DDI Start Type to Delay Start, provided the default values are used for Delay Start Delay and Delay Start Duration.
E1 Framing Mode

Parameter group

Trunk Interface

Access level

Admin

Possible values

CRC
Double

Defaults

Double

Explanation

Specifies the frame format that is to be used with E1 trunks. Select CRC for cyclic redundancy check-4 (specified in CCITT recommendation G.704). Select Double for E1 double frame format. For guidance on which parameter to set, see “Setting line code and framing mode parameters” on page 119.
E1 Hit Filter (2 ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

5 through 15 units

Defaults

5 units (10 ms)

Explanation

Specifies how long a signaling state change must be to qualify as valid. This value is adjustable in units of 2 ms. The default value sets this parameter to 10 ms.

To define this parameter, enter a number that is one-half of the desirable hit filter time. For example, to define a hit filter of 14 ms, enter 7.

WebSphere Voice Response uses this parameter only when the value of the Trunk Interface parameter is E1/A-law.
E1 Timeslot 0 Word
Parameter group

Trunk Interface

Access level

Field

Possible values

0 or 1 in each bit

Defaults

1101 1111

Explanation

If the trunk interface is E1, bit one and bits four through eight of the time slot-0 word that is in the frame that does not contain the frame alignment signal can be used by a telephone company for country-specific requirements. Set these bits to 0 or 1 in accordance with any telephone company requirements. Bits two and three are fixed at 1 and 0. The bit order is one, two,... eight.
E1 Timeslot 16 Word

Parameter group

Trunk Interface

Access level

Field

Possible values

0 or 1 in each bit

Defaults

0000 1011

Explanation

If the trunk interface is E1, bits five, seven, and eight of the time slot-16 word in frame zero (the frame containing the multiframe alignment signal) can be used by a telephone company for country-specific requirements. Set these bits to 0 or 1 in accordance with any telephone company requirements. Bits one through four and bit six are fixed at 0. The bit order is one, two,... eight.
E164 Prefixes to Strip
Parameter group
VoIP SIP Signalling

**Applicability**
DTEA and DTNA

**Access level**
Admin

**Possible values**
Prefixes separated by a comma

**Defaults**
Null

**Explanation**
Defines which leading digits should be stripped from SIP numeric user parts for E.164 country codes, for example, '1,44,393' will strip +1, +44 or +393 from the number, '+' will strip only the +, leaving the country code.
Echo Suppression Level (dBm)

Parameter group

Trunk Interface

Access level

Admin

Possible values

-76 through -31

Defaults

-50

Explanation

Specifies the energy of the echo-cancelled signal with reference to the output energy during the calibration of the echo canceller. If the energy comparison is less than or equal to the value for a consecutive 200 ms period, calibration is successful. For information about calibrating the echo canceller, see the WebSphere Voice Response for AIX: Designing and Managing State Table Applications book.
EDL Call Information After Off Hook

Parameter group

Application Server Interface

Access level

Admin

Possible values

No
Yes

Defaults

No

Explanation

Specifies whether WebSphere Voice Response receives incoming call information (for example, calling number, called number) on the exchange data link (EDL) before or after answering an incoming call.

Select Yes when the Signaling Process Type parameter (Channel Group parameter group) is set to ACL and the EDL Switch Type parameter (Exchange Data Link parameter group) is set to Siemens, or when Signaling Process Type is set to SMSI/SMDI/VMS and EDL Switch Type is set to Ericsson. This makes the channel process go off hook before attempting to get call information, rather than waiting to go off hook when it issues an AnswerCall action.

Select No when Signaling Process Type is set to SMSI/SMDI/VMS and EDL Switch Type is set to AT&T/Lucent or Northern Telecom.
EDL Communication Port
Parameter group

Exchange Data Link

Access level

Admin

Possible values

Defaults

/dev/tty2

Explanation

For SMSI, SMDI, VMS, or ACL, specify the device definition of the serial port on the pSeries computer to which the exchange data link is connected. Specify the device name as configured in SMIT. This is usually /dev/tty$n or an SMSI or VMS exchange data link or /dev/mpq$n for an ACL exchange data link, where $n is the number of the port used to create the physical exchange data link connection.

For a CallPath_SigProc link, specify the server name or Internet Protocol (IP) address of the CallPath Server system. The exact format depends on how your CallPath Server systems are set up.

Note: This parameter was called TTY or MPQ Port in previous releases.
EDL Data Rate (bits/sec)

Parameter group

Exchange Data Link

Access level

Admin

Possible values

110
300
600
1200
2400
4800
9600

Defaults

1200

Explanation

Specifies the data transfer speed of the exchange data link.

Note: This parameter was called Data Rate in previous releases.
EDL Message Info Age Limit (Seconds)

Parameter group

Application Server Interface

Access level

Admin

Possible values

3 through 15

Defaults

10

Explanation

Specifies the maximum age at which information received from the exchange data link is still thought to be valid. When information is older than the number of seconds defined by this parameter, WebSphere Voice Response assumes it is from a previous call and does not use it.
EDL Message Info Time Out (Seconds)

Parameter group

Application Server Interface

Access level

Admin

Possible values

3 through 10

Defaults

5

Explanation

Specifies how long WebSphere Voice Response waits to receive information from the exchange data link before it assumes that the information (or the link) is not available.
EDL Parity

Parameter group
Exchange Data Link

Access level
Admin

Possible values
Even
None
Odd

Defaults
Even

Explanation
Specifies the parity that is set at the switch and that is applicable to transmissions over the exchange data link.

Note: This parameter was called Parity in previous releases.
EDL Switch Type

Parameter group
Exchange Data Link

Access level
Admin

Possible values
AT&T/Lucent
Ericsson
None
Northern Telecom
Siemens Hicom

Defaults
None

Explanation
Specifies the switch type. Select:
• **AT&T/Lucent** for an SMSI exchange data link
• **Northern Telecom** for an SMDI exchange data link
• **Ericsson** for a VMS exchange data link
• **Siemens** for an ACL exchange data link.

You must also set Signaling Process Type in the Channel Group parameters group, correctly.
Enable Echo Cancellation

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA only

Access level

Admin

Possible values

No
Yes

Defaults

No

Explanation

Setting this parameter to 'yes' will enable an echo canceller to be built as part of the DSP processing resource.

Note: DTNA does not provide echo cancellation.
Enter Key

Parameter group

Application Server Interface

Access level

Admin

Possible values

# *
0 through 9

Defaults

#

Explanation

Specifies the key on the telephone keypad that can be used as an enter key. Typically, the enter key is used to indicate that the caller has finished entering data from the telephone key pad. The voice application uses the GetData action to retrieve the data. The GetData action ends when it detects that the enter key has been pressed. The default value means that callers should press # at the end of the data.
Error Table Address

Parameter group

General

Access level

Admin

Possible values

0x30000000 through 0xA0000000

Defaults

0x50000000

Explanation

Specifies the address at which WebSphere Voice Response attaches its error table shared memory segment. If you have another software package that is used in a custom server and that attaches shared memory at a fixed location that is also used by WebSphere Voice Response, you can use this parameter to specify a different location for WebSphere Voice Response to attach its segment.

The value must not be the same as the value of either Control Memory Address or Buffer Pool Address.
Errorlog Wrap Threshold (recs)

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 65535 entries

Defaults

800

Explanation

Specifies how large the WebSphere Voice Response error log file ($DB/current_dir/oamlog/errorlog) can become before wraparound occurs and records are written to the beginning of the file again. A value of 0 specifies that wraparound should never occur; records continue to be written to the end of the file, regardless of how large it gets. Ensure that you have enough space on the file system, because a full file system can seriously affect the performance of WebSphere Voice Response.

Note: Changes to this parameter do not take effect during a normal shutdown of WebSphere Voice Response. You must stop all processes, including the dtalarmd daemon by using the DT_shutdown command, then restart WebSphere Voice Response.
**Extra Channel Process**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

5 through 1000

**Defaults**

10

**Explanation**

Specifies the number of extra channel management processes that WebSphere Voice Response will create and run to ensure, for example, that calls are always answered during unusually heavy traffic periods.
File Availability Check Timeout

Parameter group

Application Server Interface

Access level

Admin

Possible values

3 through 60 seconds

Defaults

15 seconds

Explanation

If DB2 does not respond to a query correctly inside this time, a red error is logged and the defined action is taken for phone calls. If all is well, another check is made after a quarter of this time with a minimum interval of one second. After an error, the interval is one second.
Forward Key

Parameter group

Application Server Interface

Access level

Admin

Possible values

#
*
0 through 9

Defaults

9

Explanation

Specifies the key on the telephone keypad that can be used to skip forward through a voice segment, voice message, user greeting, or audio name. The number of seconds skipped is specified by the Play Skip system parameter. The default value means that callers should press 9 to skip forward.
**Frequency 1 Maximum (Hz)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

0 through 4000

**Defaults**

0

**Explanation**

With Frequency 1 Minimum, specifies the first frequency component of the call progress tone.
Frequency 1 Minimum (Hz)
Parameter group
Call Progress Tones

Access level
Field

Possible values
0 through 4000

Defaults
0

Explanation
With Frequency 1 Maximum, specifies the first frequency component of the call progress tone.
Frequency 2 Maximum (Hz)

Parameter group

Call Progress Tones

Access level

Field

Possible values

0 through 4000

Defaults

0

Explanation

With Frequency 2 Minimum, specifies the second frequency component of the call progress tone.
Frequency 2 Minimum (Hz)
Parameter group
Call Progress Tones

Access level
Field

Possible values
0 through 4000

Defaults
0

Explanation
With Frequency 2 Maximum, specifies the second frequency component of the call progress tone.
Frequency 3 Maximum (Hz)
Parameter group
Call Progress Tones

Access level
Field

Possible values
0 through 4000

Defaults
0

Explanation
With Frequency 3 Minimum, specifies the third frequency component of the call progress tone.
**Frequency 3 Minimum (Hz)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

0 through 4000

**Defaults**

0

**Explanation**

With Frequency 3 Maximum, specifies the third frequency component of the call progress tone.
FXS Start Type

Parameter group

Channel Group

Access level

Admin

Possible values

Ground Start
Loop Start

Defaults

Loop Start

Explanation

Specifies whether WebSphere Voice Response uses ground start or loop start signaling. Ground start provides positive hang-up detection without state table intervention.
G711 Voice activity det/comfort noise gen
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
Setting this parameter will enable detection of silence in the incoming voice and the resulting transmission of special short RTP packets containing silence indication.

Note: DTNA will always send RTP packets, regardless of this setting.
**G711 Packet Voice Interval (ms)**

**Parameter group**

VoIP DTEA and DTNA Media

**Applicability**

DTEA only

**Access level**

Admin

**Possible values**

10 through 200

**Defaults**

20

**Explanation**

Time between the transmission of voice packets when using G711 codec.

**Note:** DTNA always uses 20ms packet voice interval for G711 (the only codec supported for DTNA).
G729 Voice activity det/comfort noise gen
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
Setting this parameter will enable detection of silence in the incoming voice and the resulting transmission of special short RTP packets containing silence indication.
G729 Packet Voice Interval (ms)

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA only

Access level

Admin

Possible values

20 through 200

Defaults

20

Explanation

Time between the transmission of voice packets when using G729 codec.
G723 Voice activity det/comfort noise gen
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
Setting this parameter will enable detection of silence in the incoming voice and the resulting transmission of special short RTP packets containing silence indication.
G723 Data Transfer Rate
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
5.3 Kb/s (ACELP)
6.3 Kb/s (MP_MLP)

Defaults
5.3 Kb/s (ACELP)

Explanation
The data rate when using G.723 codec. (5.3 Kb/s is 12:1 compression, 6.3 Kb/s in 10:1).
G723 Packet Voice Interval (ms)
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA only

Access level
Admin

Possible values
30 through 210

Defaults
30

Explanation
Time between the transmission of voice packets when using G723 codec.
Glare Detection Time (ms)
Parameter group

Signaling Type

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 5100

Defaults

40

Explanation

Specifies how long WebSphere Voice Response tests for glare after it sends a seizure signal to the switch. Glare occurs when an incoming call appears at the same time as an outgoing call is being placed.

When WebSphere Voice Response detects glare, it immediately stops sending the seizure request signal (off-hook) and accepts the incoming call from the switch.

Note: This parameter applies to all channel associated signaling protocols except R2.
<table>
<thead>
<tr>
<th>Ground Flash (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Parameter group</strong></td>
</tr>
<tr>
<td>Signaling Type</td>
</tr>
<tr>
<td><strong>Access level</strong></td>
</tr>
<tr>
<td>Field</td>
</tr>
<tr>
<td><strong>Possible values</strong></td>
</tr>
<tr>
<td>Integer multiples of 20, in the range 0 through 5100</td>
</tr>
<tr>
<td><strong>Defaults</strong></td>
</tr>
<tr>
<td>500</td>
</tr>
<tr>
<td><strong>Explanation</strong></td>
</tr>
<tr>
<td>Specifies the length of the ground flash signal that WebSphere Voice Response sends to the switch to call a feature (such as call transfer).</td>
</tr>
<tr>
<td>Set this value as specified by the switch vendor, otherwise the signal might not be interpreted correctly.</td>
</tr>
</tbody>
</table>
Hand Shake Threshold (ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

Integer multiples of 1000, in the range 0 through 65000

Defaults

6000

Explanation

Specifies how long the digital trunk adapter waits before it starts to block cells when it has lost communication with the WebSphere Voice Response software.
Hang Up Detection
Parameter group

Signaling Type

Access level
Admin

Possible values
Cadence Detection
Constant Energy Detection
Off

Defaults
Off

Explanation
Specifies whether WebSphere Voice Response is to monitor far-end audio energy on the channel in use. Select Constant Energy Detection if the switch returns constant energy, for example, dial tone, for at least 2 seconds to indicate that the switch party has disconnected. Select Cadence Detection if the switch returns an interrupted tone, for example, busy tone, when the switch party has disconnected. For more information, see “Setting parameters for hangup tone detection” on page 112.

Continuous tone detection and cadenced tone detection are mutually exclusive. WebSphere Voice Response cannot detect a cadenced tone if this parameter is set to Constant Energy Detection, and it cannot detect a continuous tone if this parameter is set to Cadence Detection.

Note: If you select either Constant Energy Detection or Cadence Detection, WebSphere Voice Response recognizes a continuous or interrupted hang-up tone, and simulates a far-end hang-up condition to the application, whether or not it is using a channel associated signaling protocol that provides a unique disconnect signal.
Hook Flash (ms)

Parameter group

Signaling Type

Access level

Admin

Possible values

Integer multiples of 20, in the range 0 through 5100

Defaults

500

Explanation

Specifies the duration of the hook flash (on-hook) signal that WebSphere Voice Response sends to the switch to call a feature (such as call transfer).

Set this value as specified by the switch vendor, otherwise the signal might be ignored (if too short) or interpreted as an on-hook signal (if too long). For example, for the ROLM 9751 switch, this value is normally set to 300 for the E&M signaling protocol and to 500 for the FXS Loop Start protocol.
Idle Channel Code
Parameter group
Trunk Interface

Access level
Field

Possible values
E1
UK/Italy
T1

Defaults
T1

Explanation
Specifies the 8-bit PCM word that is transmitted to indicate silence in a voice channel. The word for T1 is 1111 1111 (FFh). The word for E1 is 0101 0101 (55h). The word for UK/Italy is 0101 0100 (54h). This word is normally specified by the country telecommunications authority.

Ignore replaces option for Attended Transfer
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
No
Yes
Defaults
No

Explanation
Specifies whether or not the presence of an inbound REPLACES header should be used to control attended transfers. If set to No, attended transfer will not proceed unless a REPLACES header has been received inbound. If set to Yes, attended transfer will continue regardless of a REPLACES header being received inbound.

Refer to “VoIP SIP attended transfer” in WebSphere Voice Response for AIX: Voice over IP using Session Initiation Protocol for more information on attended transfer.
Inbound Call Channel Allocation Method

Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
Allocate calls from highest available channel (using 'linear' allocation)
Allocate calls using 'round robin' method
Allocate calls for D2IS
Allocate calls balanced across trunks

Defaults
Allocate calls from highest available channel

Explanation
This determines how WebSphere Voice Response allocates an incoming SIP call to a trunk and channel.

The default is by searching for the first free channel starting from the highest numbered channel and working downwards (from Trunk 16, channel 30 to Trunk 1 Channel 1). For some systems which use CTI (Computer Telephony Integration) this means that highest-numbered channels are reused very quickly and also there is an uneven spread of active channels across the complete range.

For more even channel allocation, the Allocate calls using 'round robin' search method begins at the next lower-numbered channel from the one last allocated (wrapping first to last).

To ensure the call load is spread over the adapters on the system, specify Allocate calls balanced across trunks. You must use this method if using iLBC compression.

The Allocate calls for D2IS method is for use with the Genesys supplied D2IS custom server. The allocated trunk and channel are based on the dialled
number. The first three digits of the dialled number are ignored. All digits after this are converted into a trunk and channel number. The trunk is derived from the remaining digits of the dialled number divided by 100, the channel is the remaining digits of the dialled number, modulo 100. For example:

- 4440205 is trunk 2, channel 5
- 444000205 is trunk 2, channel 5
- 1110430 is trunk 4, channel 30

**Note:** Using the 'D2IS' method, it is not possible to specify an application profile number to state table mapping. However, Number-to-application (NumToApp) mappings are possible if the first three digits map to a particular application, for example, 123001001 could be defined as 123* and 987001001 could be defined as 987* in NumToApp mappings.
Inbound DTMF Method Override

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTNA only

Access level

Admin

Possible values

Use negotiated values
DTMF using tones (‘in-band’ audio tones in the RTP stream)
DTMF using payload (RFC2833 payload)
DTMF using SIP info (using the SIP INFO method)

Defaults

Use negotiated values (what is agreed as part of the SIP protocol)

Explanation

Normally, the DTMF method is limited to RTP methods only (in-band/payload). However, using the System Parameter Inbound DTMF Method Override, it is possible to force the DTMF transmission type to be in-band, payload or (the new option) SIP Info method. In this case, the State Table ‘GetData’ action will allow collection of DTMF keys sent using the SIP INFO method. If you want the DTMF transfer method to be agreed as part of the SIP call setup process (by the SDP in the INVITE and 200 OK messages), set this option to Use Negotiated Values. See also “Outbound DTMF Method Override” on page 412.
Incoming Address Register Type
Parameter group

Signaling Type

Access level

Admin

Possible values

Fixed Length
Feature Group D

Defaults

Fixed Length

Explanation

Specifies the type of address register. The Fixed Length value selects a fixed length register, with the length defined by the Register Length system parameter. Feature Group D selects the unique protocol Exchange Access North American Signaling described in the Bellcore publication TR-NPL-000258. That is, WebSphere Voice Response collects the information field (ANI) first then collects the address field (DNIS). WebSphere Voice Response does not support any other collection sequence.
Incoming Address Signaling Type
Parameter group

Channel Group

Access level
Admin

Possible values
DTMF
MFR1
MFR1 Modified
Dial Pulse

Defaults
DTMF

Explanation
Specifies the type of address signaling that is used for incoming calls on all channels in the channel group.

If the Incoming Address Register Type parameter for the Signaling Type used by this channel group is set to Fixed, you should select **DTMF**, **MFR1**, or **Dial Pulse**. If the Address Register Type parameter is set to Feature Group D, you must select **MFR1**.
### Incoming Guard Time (ms)

**Parameter group**

- Signaling Type

**Access level**

- Field

**Possible values**

0 through 5100

**Defaults**

0

**Explanation**

Specifies how long WebSphere Voice Response waits before acknowledging a new incoming call after both the switch and WebSphere Voice Response have returned to idle (on-hook) following a previous call (incoming or outgoing) on that channel.

The Incoming Guard Time must be equal to, or less than, the Outgoing Guard Time.
Interdigit Pause Receive (ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

20 through 2047

Defaults

280

Explanation

Specifies the minimum amount of time WebSphere Voice Response expects to elapse between two dialed digits.
Interdigit Pause Transmit (ms)
Parameter group
Trunk Interface

Access level
Field

Possible values
20 through 2047

Defaults
1000

Explanation
Specifies how much time elapses between each dialed digit when WebSphere Voice Response transmits to the switch.
Interval for Checking MWI Status (s)

Parameter group
Exchange Data Link

Access level
User

Possible values
1 through 600 seconds

Defaults
60 seconds

Explanation
If “MWI Automatically Set” is set to “Yes”, then WebSphere Voice Response checks each mailbox for likely MWI activation or deactivation at the interval defined by the parameter.
IP Address

**Parameter group**

VoIP Media - Adapters

**Access level**

Admin

**Possible values**

0.0.0.0 through 255.255.255.255

**Defaults**

0.0.0.0

**Explanation**

Records the IPv4 IP address of a DTEA card. There are four instances of this parameter, one for each card.
**ISDN - Redial Limitation**

**Parameter group**
Trunk Interface

**Access level**
Field

**Possible values**
No
Yes

**Defaults**
No
Yes in Japan

**Explanation**

The Redial Limitation facility allows the system to prevent calls to an individual number after a certain number of failures.

If this parameter is set to Yes, then the redial limitation functionality is enabled; using the parameters in the ISDN Signaling group in system configuration.
<table>
<thead>
<tr>
<th><strong>Parameter group</strong></th>
<th><strong>Trunk Interface</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Access level</strong></td>
<td>Field</td>
</tr>
<tr>
<td><strong>Possible values</strong></td>
<td>No</td>
</tr>
<tr>
<td></td>
<td>Yes</td>
</tr>
</tbody>
</table>

**Defaults**
No

**Explanation**
Specifies that ISDN non-facility associated signaling (NFAS) is to be used. If you select Yes, you also need to specify an ISDN Trunk Identifier.
ISDN Transfer Type

Parameter group
Trunk Interface

Access level
Admin

Possible values
None
2B Channel
RLT

Defaults
None

Explanation
This parameter allows you to select a transfer type for ISDN. Transfer is supported only on the Nortel DMS-100 and DMS-250 switches. RLT is supported on the DMS-250. Both RLT and 2 B-Channel Transfer are supported on the DMS-100.
**ISDN Trunk Identifier**

**Parameter group**

Trunk Interface

**Access level**

Admin

**Possible values**

0 through 31

**Defaults**

0

**Explanation**

Identifies the trunk if the ISDNT1-NFAS Support parameter is set to YES. Specify the value that identifies the trunk in your switch configuration.
**L2 - Link Handshake Timer T203 (ms)**

**Parameter group**

ISDN Signaling

**Access level**

Field

**Possible values**

Integer multiples of 5, in the range 5000 through 20000

**Defaults**

10000

Specifies the value of the T203 timer, which acts as a handshake mechanism when ISDN L2 is in Multiframe Established state or Active state. If the timer expires, an L2 message is sent to the switch to ensure that it is still active. If no response is received, L2 comes out of the active state and tries to restart the L2 trunk.
L2 - Link Release Timer T200 (ms)

Parameter group
ISDN Signaling

Access level
Field

Possible values
Integer multiples of 5, in the range 700 through 2000

Defaults
1000

Explanation
Specifies the value of the T200 timer, which acts as a retry time when attempting to establish the L2 data link. On starting to establish the link, the timer is started. The timer is stopped when the switch responds. On expiry, another L2 link establish message is sent to the switch and the timer is restarted. This is done for a particular number of retries until it is assumed that the switch is not ready to activate L2.
L3 - T309 Support (ms)

Parameter group
ISDN Signaling

Access level
Field

Possible values
No
Yes

Defaults
Yes

Explanation
Specifies whether the ISDN L3 T309 timer is enabled. If you select Yes, active calls are not immediately cleared when a temporary failure in the ISDN signaling channel (D-channel) occurs. Calls being set up or cleared are lost.

If you select No, all calls are immediately cleared whenever any D-channel failure occurs.
L4 - Called/Calling Party Numbering Plan

Parameter group

ISDN Signaling

Access level

Admin

Possible values

ISDN
National Standard
Private
Unknown

Defaults

Unknown

Explanation

Specifies the telephone numbering scheme that is used in the network. The most common for ISDN is the ISDN E.164 plan. If the plan is not known, select Unknown. This is normally acceptable.

Note: This parameter applies to all calls that are made through ISDN, and is not set on a call-by-call basis.
L4 - Called/Calling Party Numbering Type

Parameter group
ISDN Signaling

Access level
Admin

Possible values
Abbreviated Number
International Number
National Number
Network-Specific Number
Subscriber Number
Unknown

Defaults
Unknown

Explanation
Specifies the format that is used for calling and called party numbers sent on an ISDN link. Select the value that matches the numbering type that is in use on the switch.

Note: This parameter applies to all calls made using ISDN and is not set on a call-by-call basis.
L4 - Facility Timeout (s)

**Parameter group**

ISDN Signaling

**Access level**

Field

**Possible values**

1 through 60

**Defaults**

10

**Explanation**

Specifies the maximum time to wait for a response to a facility message. If this timer expires, it is assumed that a response to the facility message is not going to occur, and suitable action is taken.
L4 - Facility Transfer Completion Timeout (s)

Parameter group
ISDN Signaling

Access level
Field

Possible values
1 through 60

Defaults
10

Explanation
Specifies the maximum time to wait for the switch to complete a transfer operation after a transfer facility response is received from the switch. Normally, the switch would start clearing the two calls that are in the transfer before this timer expires. If the switch has not started to clear the two calls, ISDN L4 clears them.
Level 1 Maximum (dBm)
Parameter group
Call Progress Tones

Access level
Field

Possible values
-43 through 8

Defaults
0

Explanation
With Level 1 Minimum, defines the level of the first frequency component of the call progress tone.
Level 1 Minimum (dBm)
Parameter group
Call Progress Tones

Access level
Field

Possible values
-43 through 8

Defaults
0

Explanation
With Level 1 Maximum, defines the level of the first frequency component of the call progress tone.
Level 2 Maximum (dBm)

Parameter group

Call Progress Tones

Access level

Field

Possible values

-43 through 8

Defaults

0

Explanation

With Level 2 Minimum, defines the level of the second frequency component of the call progress tone.
**Level 2 Minimum (dBm)**

Parameter group

Call Progress Tones

**Access level**

Field

**Possible values**

-43 through 8

**Defaults**

0

With Level 2 Maximum, defines the level of the second frequency component of the call progress tone.
**Level 3 Maximum (dBm)**

**Parameter group**
Call Progress Tones

**Access level**
Field

**Possible values**
-43 through 8

**Defaults**
0

**Explanation**
With Level 3 Minimum, defines the level of the third frequency component of the call progress tone.
**Level 3 Minimum (dBm)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

-43 through 8

**Defaults**

0

**Explanation**

With Level 3 Maximum, defines the level of the third frequency component of the call progress tone.
License Request Timeout (seconds)
Parameter group
General

Access level
Admin

Possible values
10 through 65535 seconds

Defaults
180

Explanation
This option allows you to select the timeout value, in seconds, that is used by the License Use Manager. Requested licences are held for this period before being automatically released by the manager. WebSphere Voice Response refreshes the licence before the period expires. The default is 3 minutes.
Line Identifier Number Length
Parameter group

Exchange Data Link

Access level

Admin

Possible values

1 through 64

Defaults

7

Explanation

Specifies the number of characters in each Message Info Line Identifier. You set the actual line identifiers by using the Message Info Line Identifier parameter in the Channel parameters group. Switches that provide an exchange data link use these line identifiers to identify the channels.

For SMDI or SMSI, select a value of 7. For VMS, select a value from 2 to 5. For ACL, select a value of 6. For details of the formats of the Line Identifiers, see “Message Info Line Identifier” on page 388.
**Low Channel Process Clear Threshold**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

2 through 500

**Defaults**

16

**Explanation**

Specifies the threshold for alarm 2008. When the number of free channel processes rises above the number specified (after falling below the Low Channel Process Warning Threshold), green alarm 2008 is written to the error log and displayed in the System Monitor. The number that is specified must be higher than the number that is specified for the Low Channel Process Warning Threshold parameter.

To increase the number of channel processes available, increase the value of the Extra Channel Process parameter.
Low Channel Process Warning Threshold

Parameter group

Application Server Interface

Access level

Admin

Possible values

1 through 500

Defaults

10

Explanation

Specifies the threshold for alarm 2007. When the number of free channel processes drops below the number specified, yellow alarm 2007 is written to the error log and displayed in the System Monitor. The number that is specified must be lower than the number that is specified for the Low Channel Process Clear Threshold parameter.

To increase the number of channel processes available, increase the value of the Extra Channel Process parameter.
Maintenance Message Protocol Discriminator

Parameter group

ISDN Signaling

Access level

Admin

Possible values

0×03
0×43

Defaults

0×43

Explanation

This selects the value of the Maintenance Message Protocol Discriminator. This value is used in every maintenance message to identify the message as a maintenance message to the network. Some networks use a different value for this parameter. Change the parameter to match the value specified by your network provider.
**Map**

**Parameter group**

Key Signals

**Access level**

Admin

**Possible values**

0 through 9
A through D
#
*
blank

**Defaults**

(Each entry has a different default value.)

**Explanation**

Specifies the character that is assigned to a key on the telephone key pad. For more details see [“Key signals parameter group” on page 188](#).
Max Number of Screens Saved by 3270 Exec
Parameter group

Application Server Interface

Access level
Admin

Possible values
0 through 3000

Defaults
1000

Explanation
Specifies the maximum number of screens that the Save Screen script language statement can save into the WebSphere Voice Response database. This number applies to the Save Screen statements in all servers. That is, if the value is set to 500, when 500 screens have been saved, no more screens can be saved by any servers.
Maximum Cached Buffers

Parameter group

Application Server Interface

Access level

Admin

Possible values

10 through 44000

Defaults

300

Explanation

Specifies the maximum number of buffers that are allocated to cache voice segments that are used by prompts. The cache can grow above this size but, when possible, segments in the cache are discarded to keep below this limit. This parameter should *never* be set to more than about two-thirds of the value of the Number of Pool Buffers parameter.
Maximum Dial Tone Wait (Seconds)

Parameter group

Application Server Interface

Access level

Admin

Possible values

1 through 10

Defaults

2

Explanation

Specifies the maximum amount of time a voice application waits for the dial tone when the application state table is running a Dial, MakeCall, or TransferCall action.
**Maximum MPN Digits**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

32 through 4000

**Defaults**

136

**Explanation**

Specifies the maximum number of digits, including those before and after the decimal point, in a multiple-precision number that is the result of a constant assignment or an addition, subtraction, or multiplication.

**Note:** The maximum number of digits to the right of the decimal place in a multiple-precision number that is the result of a division, is specified by the System : MPN : Maximum decimal places variable.
**Maximum Playback Level (dBm)**

**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

-21 through 0

**Defaults**

-15

**Explanation**

Specifies the maximum audio level that is played to a caller. The process WebSphere Voice Response uses to determine the volume at which a voice segment is played is as follows:

1. When a voice segment is stored in the database (see the *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* book for details), WebSphere Voice Response determines the maximum signal level.

2. Just before a voice segment is played, WebSphere Voice Response calculates the difference between Maximum Playback Level and the maximum level that is in the segment.

3. When it plays it to a caller, WebSphere Voice Response attenuates or amplifies the whole voice segment by the calculated difference.

Here are two examples:

- The maximum signal power in the voice segment is -18 dBm and Maximum Playback Level is set to -15 dBm. The difference is -15 - (-18) = +3dB. The whole voice segment is therefore amplified by 3 dB. Note that those parts of the segment that are less than -18 dBm are amplified by 3 dB, but are not increased to -15 dBm. For example, signals at -23 dBm are increased to -20 dBm.

- The maximum signal power in the voice segment is -10 dBm and Maximum Playback Level is set to -15 dBm. The difference is -15 - (-10) = -5dB. The whole voice segment is therefore attenuated by 5 dB. For example, signals at -14 dBm are decreased to -19 dBm.
Therefore, the signal power level for a voice segment never exceeds the value that is specified by Maximum Playback Level. This technique also ensures that all voice segments are played at the same level regardless of the level at which they were recorded. Note, however, that because the same level adjustment applies to the entire segment and is based on the maximum signal level in the whole segment, a burst of loud voice (or other sound) in the segment can make the remainder of the segment sound quieter than expected.

WebSphere Voice Response continuously adjusts the gain of the connection, to prevent the level from exceeding this value. The value set for this parameter is the maximum transmit level that is allowed by the country telecommunications authority.
Maximum Retries for Pack/DTTA Reenabling

Parameter group

General

Access level

Admin

Possible values

1 through 20

Defaults

5

Explanation

Specifies the maximum number of times in one hour that WebSphere Voice Response automatically re-enables DTTA following an irrecoverable error.
Maximum Ring Time (Seconds)
Parameter group
Application Server Interface

Access level
Admin

Possible values
0 through 120

Defaults
20

Explanation

Specifies the maximum amount of time a voice application waits to detect an answer after it has performed a Dial, MakeCall, or TransferCall action.

The value of Maximum Ring Time is used if the state table action Ring Time parameter is set to its default value of 0. The state table action Ring Time parameter value is used if its value is greater than 0, whatever the value of the Maximum Ring Time parameter.

If the state table action parameters Ring Time and Ring Wait are both set to 0, the action returns the Succeeded result immediately after it has sent the last digit in the digit string (that is, the action does not wait for any more activity on the channel). This action is known as a blind dial, blind makecall, or blind transfer.

This information is summarized in Table 18

Table 18. How Ring Time and Ring Wait parameters affect answer detection

<table>
<thead>
<tr>
<th>System parameter</th>
<th>State Table action parameter</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Ring Time</td>
<td>Ring Time</td>
<td>Ring Wait</td>
</tr>
<tr>
<td>Any value</td>
<td>0</td>
<td>&gt;0</td>
</tr>
<tr>
<td>Any value</td>
<td>&gt;0</td>
<td>Any value</td>
</tr>
<tr>
<td>System parameter</td>
<td>State Table action parameter</td>
<td>Result</td>
</tr>
<tr>
<td>---------------------</td>
<td>------------------------------</td>
<td>------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Maximum Ring Time</td>
<td>Ring Time 0</td>
<td>Action returns the Succeeded result immediately after sending the last digit in the digit string</td>
</tr>
<tr>
<td>Any value 0</td>
<td>0</td>
<td>Action returns the Succeeded result immediately after sending the last digit in the digit string</td>
</tr>
</tbody>
</table>
Maximum Ring Wait (Seconds)

Parameter group
Application Server Interface

Access level
Admin

Possible values
0 through 30

Defaults
15

Explanation
Specifies the maximum amount of time a voice application waits for a ringback tone after it has sent a digit string, when the application state table is running a Dial, MakeCall, or TransferCall action.

The value of Maximum Ring Wait is used if the state table action Ring Wait parameter is set to its default value of 0. The state table action Ring Wait parameter value is used if its value is greater than 0, whatever the value of the Maximum Ring Wait parameter.

If the state table action parameters Ring Time and Ring Wait are both set to 0, the action returns the Succeeded result immediately after it has sent the last digit in the digit string (that is, the action does not wait for any more activity on the channel). This action is known as a blind dial, blind makecall, or blind transfer.

This information is summarized in Table 19 on page 383.

To guarantee detection of ringback tone, the value for Maximum Ring Wait must be at least twice the ringing cycle time. For example, if the ringing cadence is 1 second on, 3 seconds off, set Maximum Ring Wait to 8 seconds or more.

Note: When using the U.S./Canada Faster Transfer Call Progress Tone template, WebSphere Voice Response returns a Succeeded result when the ringback tone frequencies are recognized; it does not wait to recognize the cycle time.
Table 19. How Ring Time and Ring Wait Parameters Affect Ringback Tone Detection

<table>
<thead>
<tr>
<th>System parameter</th>
<th>State Table action parameter</th>
<th>Ring Time</th>
<th>Ring Wait</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Ring Wait</td>
<td></td>
<td>&gt;0</td>
<td>0</td>
<td>Use Maximum Ring Wait value</td>
</tr>
<tr>
<td>Any value</td>
<td></td>
<td>Any value</td>
<td>&gt;0</td>
<td>Use Ring Wait value</td>
</tr>
<tr>
<td>Any value</td>
<td></td>
<td>0</td>
<td>0</td>
<td>Action returns the Succeeded result immediately after sending the last digit in the digit string</td>
</tr>
</tbody>
</table>

Appendix A. System parameters 383
Maximum Silence Duration (ms)

Parameter group

Trunk Interface

Access level

Admin

Possible values

Integer multiples of 1000, in the range 3000 through 40000

Defaults

12000

Explanation

Specifies how long a Record... action waits after the speaker stops speaking. If the speaker does not begin to speak again before this period elapses, the Record... action ends. However, if the caller does not start to speak, the “period of silence detection” does not start.

This parameter applies to:

- Voice messages
- Audio names
- Greetings that are being recorded by callers
- Voice segments that are being recorded for a voice application to use

However, the “period of silence detection” does not start while anything is being played.

Note: With the Subscriber Loop (SL) protocol, WebSphere Voice Response ends a call when the silence detection timeout occurs. For other signaling protocols, WebSphere Voice Response cannot end a call through silence detection.
Maximum Silence Level (0.5 dBm)

Parameter group

Trunk Interface

Access level

Field

Possible values

-90 through -110 units

Defaults

-99 units (-49.5 dBm)

Explanation

Specifies the maximum expected level of audio input that is interpreted as silence. Any audio input above this level is interpreted as speech.

Enter a number for this parameter that is twice the required maximum silence level. For example, if the level desired is -48.5 dBm, define the value of this parameter as -97.
**Message Header Format**

**Parameter group**
VoIP SIP Signaling

**Access level**
Admin

**Possible values**
Compact
Long

**Defaults**
Long

**Explanation**
Determines the format of the message headers when a message is created.

Headers can be long, which is the default, or compact. Using compact headers reduces the size of the message.
**MWI Trunk**

**Parameter group**

ISDN Signaling

**Access level**

Field

**Possible values**

Comma-delimited list of digits.

**Defaults**

0

**Explanation**

Comma-delimited list of one or more trunks to which the Message Waiting Indicator (MWI) requests are to be sent.

With the default setting (0), MWI requests are sent to the first enabled ISDN trunk found that is not in an alarm state.

To send MWI requests to multiple trunks, change the value of the MWI Trunk parameter to a comma-delimited list of trunks to which the MWI requests are to be sent. Any MWI request will be sent to all trunks specified in the list.

The following are three examples of trunk lists for use in the MWI Trunk parameter:

- "1,3"
- "2,4,5"
- "1"
**Message Info Line Identifier**

**Parameter group**

Channel

**Access level**

Admin

**Possible values**

A character string

**Defaults**

TEST101

**Explanation**

Specifies the code that the switch uses to identify this channel when it transmits information via the exchange data link about the call that is in process on this channel.

The length of the character string must be equal to the length specified by the Line Identifier Number Length parameter in the Exchange Data Link parameters group.

The formats of the line identifiers are different for the various exchange data links:

**ACL**  
The line identifier is an ASCII character, usually Q for an analog tie trunk, followed by five hexadecimal numbers in the format \( TL \) where \( TT \) are two hexadecimal digits that specify the trunk group, usually in the range 01 to FF; and \( LLL \) are three hexadecimal digits that specify the line number within a trunk group, usually in the range 001 to FFF.

**SMDI or SMSI**  
The line identifier is made up of seven digits in the format \( ggnnnn \), where \( ggg \) is a three-digit number that specifies the message desk number (also referred to as the line group), usually in the range 001 to 999; and \( nnnn \) is a four digit-number that specifies the message desk terminal (also referred to as the line number), usually in the range 0001 to 2047.

**VMS**  
The line identifier is a number between two and five digits in length.
For a specific implementation, the number of digits is fixed (that is, all lines from a switch have the same number of digits).

**CallPath**

The line identifier is the physical phone number for the channel.
CallPath refers to this as the Party number.

Each line identifier that is configured in the WebSphere Voice Response system must be unique and must be configured on the correct channel. Otherwise call information might be sent to the wrong channel, resulting in the call information either being sent to the wrong incoming call, or being lost.

You might have to ask your service provider to give details of all the line identifiers that have been set up on your switch. The numbers do not always follow a logical sequence from one channel to the next, so it is not safe to assume that the identifiers just increment from the first one.
MFR1 Receive Level (0.5 dBm)

Parameter group

Trunk Interface

Access level

Field

Possible values

-60 through 0 units

Defaults

-50 units (-25 dBm)

Explanation

Specifies the detection threshold for all multifrequency (MF) tones that are received by WebSphere Voice Response.

For this parameter, enter a number that is twice the required level. For example, if the level desired is -48.5 dBm, define the value of this parameter as -97.
MFR1 Stop Key

Parameter group

Trunk Interface

Access level

Field

Possible values

700/1700 Hertz
900/1700 Hertz
1300/1700 Hertz
1500/1700 Hertz

Defaults

1500/1700 Hertz

Explanation

Specifies the MFR1 tone to be used for the stop key when MFR1 address signaling is used.

The default value corresponds to the ST (end of pulsing) signal in ITU-T Recommendation Q.323 and Bellcore Technical Reference TR-NWT-000506.
Minimum Speech Level (0.5 dBm)

Parameter group
Trunk Interface

Access level
Field

Possible values
-20 through -81 units

Defaults
-81 (-40.5 dBm)

Explanation
Specifies the minimum expected level of audio input. Any audio input below this level is interpreted as silence.

For this parameter, enter a number that is twice the required minimum speech level. For example, if the level desired is -38.5 dBm, define the value of this parameter as -77.
Music Absolute Silence Threshold (dBm)

Parameter group

Trunk Interface

Access level

Field

Possible values

-75 through 0

Defaults

-50

Explanation

Specifies the lowest audible volume for background music. When the absolute level of background music falls below this level, WebSphere Voice Response unlinks the music channel from the telephony channel. (If the value is set too high, callers might hear a sudden end to the music.)
Music Automatic Fade Before Actions

Parameter group

Application Server Interface

Access level

Admin

Possible values

No
Yes

Defaults

Yes

Explanation

Specifies whether the system fades out the background music before one of the following actions: GetKey, GetData, GetFindData, GetFindName, GetPassword, GetText, RecordVoiceSegment, RecordAudioName, RecordUserGreeting, RecordVoiceMessage, RecordVoiceToHost. Be aware that if you set this parameter to No, an echo of the background music might be recorded.

This parameter can be overridden on an application basis by the SV226 system variable.

Note: The background music always fades out before TransferCall, MakeCall, AnswerCall, and RecognizeWord, whatever the value of this parameter.
Music Automatic Fade Time Default (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

Integer multiples of 100, in the range 0 through 10 000

Defaults

1000 (1 second)

Explanation

Specifies the length of time WebSphere Voice Response takes to fade out the background music for a GetKey, GetData, GetFindData, GetFindName, GetPassword, GetText, RecordVoiceSegment, RecordAudioName, RecordUserGreeting, RecordVoiceMessage, or RecordVoiceToHost action.

This parameter can be overridden on an application basis by the SV225 system variable.

Note: The background music always fades out before TransferCall, MakeCall, AnswerCall, and RecognizeWord, whatever the value of this parameter.
Music Channels Maximum
Parameter group

Trunk Interface

Access level
Admin

Possible values
0 through 8

Defaults
1

Explanation

Specifies the maximum number of music channels that the telephony channels on this trunk can use at the same time. The default allows all applications to use the same music from one music channel. To enable applications to have access at the same time to more than one piece of music, you must increase the maximum number of music channels, but be aware that by using more music channels, you decrease system performance.
Music Volume Ceiling Default (dBm)

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 50

Defaults

14

Explanation

Specifies the difference, in decibels (dBm), between the maximum permissible volume on the line and the maximum volume at which background music is played. If you make the difference smaller, background music can be played louder. For an example of how this would be used, see “Prompt Volume Ceiling Default (dBm)” on page 431.

WebSphere Voice Response ensures that the values of Prompt Volume Ceiling Default (dBm) and Music Volume Ceiling Default (dBm) cannot together cause the volume on the line to exceed the maximum permissible volume.

This parameter can be overridden on an application basis by the SV224 system variable.
MWI Automatically Set
Parameter group
Exchange Data Link

Access level
Admin

Possible values
Yes
No

Defaults
No

Explanation
Specifies whether the Message Waiting Indicator is automatically set whenever a message arrives.
**MWI Number Length**

**Parameter group**
Exchange Data Link

**Access level**
Admin

**Possible values**
1 through 64

**Defaults**
7

**Explanation**
Specifies the number of digits in the MWI number that WebSphere Voice Response sends over the exchange data link. This is used only if MWI Number Padding is set to Yes.
MWI Number Padding
Parameter group
Exchange Data Link

Access level
Admin

Possible values
Yes
No

Defaults
No

Explanation
Specifies whether WebSphere Voice Response pads any numbers in a message waiting indication (MWI) request. If you select No, WebSphere Voice Response does not pad the numbers. Any numbers received are handled as the actual MWI number.

If you select Yes, all numbers that are sent to the exchange data link are padded with a leading ASCII character. The number is padded with the character that is defined by MWI Number Padding Character to make it the length that is defined by MWI Number Length. If the length of the MWI number that is received from the signaling library is greater than the MWI Number Length parameter, an error is issued.

The MWI Number Padding parameter allows variable length numbers to be sent to an exchange data link protocol that supports only fixed length numbers.
MWI Number Padding Character

Parameter group

Exchange Data Link

Access level

Admin

Possible values

Alphanumeric character

Defaults

0 (ASCII zero)

Explanation

Specifies the character that WebSphere Voice Response uses to pad the MWI number to form a fixed length string containing the number. This parameter is used only if the MWI Number Padding parameter is set to Yes.
No Answer Warning (ms)

Parameter group

Signaling Type

Access level
Admin

Possible values
20 through 180000

Defaults
120000

Explanation
This parameter applies to channel associated signaling only. Specifies the maximum amount of time allowed for WebSphere Voice Response to answer an incoming call when the call is qualified. If WebSphere Voice Response does not answer in this time, an error, 17038 (signal detected but could not establish call) is logged. No other action is taken. Each time the error is reported, WebSphere Voice Response starts the timing again. This action can continue for an unlimited time.
Normal Play/Record Max Data (KBytes)

Parameter group

Application Server Interface

Access level

Admin

Possible values

32 through 64

Defaults

32

Explanation

Specifies the number of memory blocks (mbufs). Select a higher value to reduce the number of write underruns that occur when concatenated voice segments are played. Be aware that mbufs are shared with other programs, particularly network services.
Number of 3270 Exec Processes to Spawn

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 254

Defaults

32

Explanation

Specifies how many 3270 executor processes WebSphere Voice Response starts. A 3270 executor process manages 3270 terminal sessions. The recommended value is one executor for every 3270 session on which you expect callers to access a voice application that uses a 3270 server.
Number of Nak Retries

Parameter group
Exchange Data Link

Access level
Admin

Possible values
0 through 10

Defaults
2

Explanation
Specifies how many times WebSphere Voice Response tries to make contact with the switch over the exchange data link when the attempts are not being acknowledged. After this number of unacknowledged attempts, WebSphere Voice Response assumes that the exchange data link is not working.
Number of Non Swap State Tables

Parameter group
Application Server Interface

Access level
Admin

Possible values
0 through 65535

Defaults
5

Explanation
Specifies the maximum number of voice application state tables that can be fixed in memory. A state table that is fixed in memory remains there until WebSphere Voice Response is shut down. The system does not swap it out.
**Number of Pool Buffers**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

50 through 64000

**Defaults**

500

**Explanation**

Specifies the total number of pool buffers. The buffer pool is used internally for communication between WebSphere Voice Response processes and also for the voice segment cache. (The number of these buffers allowed for the cache is set by the Maximum Cache Buffers parameter, which should never be set to more than two thirds of the total number of pool buffers.)

Be careful when setting this parameter; do not set it higher than necessary. Each buffer contains 4K bytes, and you need enough real memory free in the system for every buffer. For example, setting this parameter to 2000 means that you must have 8M bytes of memory free.

If you set this parameter too high, you might not have enough disk space to start WebSphere Voice Response. When this happens, you cannot log on to reset this parameter, but you can use the RDSETBUFS command-line utility to reset it. To use this utility, type RDSETBUFSn, where n is the total number of pool buffers. RDSETBUFS also resets the value of the Maximum Cached Buffers parameter to no more than two thirds of the total number of pool buffers.
Number of VAGSERVERs

Parameter group

General

Access level

Admin

Possible values

1 through 16

Defaults

1

Explanation

Specifies how many VAGSERVER processes you want WebSphere Voice Response to start. You can improve the responsiveness of voice messaging applications by increasing the value of this system parameter. This is especially noticeable on SMP systems and on clients of a single system image.
Number of Voice Messaging Servers

Parameter group

General

Access level

Admin

Possible values

1 through 8

Defaults

1

Explanation

Specifies how many voice messaging server processes you want WebSphere Voice Response to start. You can improve the responsiveness of voice messaging applications by increasing the value of this system parameter. This is especially noticeable on SMP systems and on clients of a single system image.
Operating Status
Parameter group
Trunk Interface

Access level
Admin

Possible values
Available
Defined
Enabled
Inservice

Defaults
Trunk 1:
Available
All other trunks:
Defined

Explanation
Specifies the initial operating status of the pack as follows:

Available
The digital trunk adapter is present but is not ready to communicate with the trunk. The trunk is not ready to process calls. Use the System Monitor from the Operations menu to enable the pack and set the channels Inservice.

Defined
The pack and associated trunk interface are fully disabled. Defined is a place holder until the digital trunk adapter is physically installed in the pSeries computer.

Enabled
The required microcode is loaded and diagnostics are run. The pack cannot be used, however, until you put the relevant channels In Service by using the System Monitor option on the Operations menu.

Inservice
The required microcode is loaded and diagnostics are run (this takes about a minute) and the pack is ready to process calls. The channels are set to Inservice: ready to make or receive calls. This is the normal setting when you are using the system in production.
Organization Name
Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
A character string

Defaults
Null

Explanation
If this field is completed, the value can be added to the call setup messages, to identify the group from which the call is being made.
Outbound DTMF Method Override
Parameter group

VoIP DTEA and DTNA Media

Applicability

DTNA only

Access level

Admin

Possible values

Use negotiated values
DTMF using tones (‘in-band’ audio tones in the RTP stream)
DTMF using payload (RFC2833 payload)
DTMF using SIP info (using the SIP INFO method)

Defaults

Use negotiated values (what is agreed as part of the SIP protocol)

Explanation

Normally, the DTMF method is limited to RTP methods only (in-band/payload). However, using the System Parameter Outbound DTMF Method Override, it is possible to force the DTMF transmission type to be in-band, payload or (the new option) SIP Info method. In this case, any State Table ‘Dial’ action will cause DTMF keys to be sent using the SIP INFO method. If you want the DTMF transfer method to be agreed as part of the SIP call setup process (by the SDP in the INVITE and 200 OK messages), set this option to Use Negotiated Values. See also “Inbound DTMF Method Override” on page 342.
Outbound SIP INFO

Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
application/vnd.nortelnetworks.digits
application/dtmf-relay

Defaults
application/vnd.nortelnetworks.digits

Explanation
The format that is used to send outbound SIP INFO header information for DTMFs initiated by WebSphere Voice Response.
Outgoing Address Register Type

Parameter group

Signaling Type

Access level

Admin

Possible values

Fixed Length
Feature Group D

Defaults

Fixed Length

Explanation

Specifies the type of address register. The Fixed Length value selects a fixed length register, with the length defined by the Register Length system parameter. Feature Group D selects the unique protocol Exchange Access North American Signaling described in the Bellcore publication TR-NPL-000258. That is, WebSphere Voice Response sends the information field (ANI) first then sends the address field (DNIS). WebSphere Voice Response does not support any other sending sequence.
**Outgoing Address Signaling Type**

*Parameter group*

Channel Group

**Access level**

Admin

**Possible values**

DTMF
MFR1
Dial Pulse

**Defaults**

DTMF

**Explanation**

Specifies the type of address signaling that WebSphere Voice Response uses when it makes an outgoing call. The specified value applies to all channels in the channel group.
**Outgoing Guard Time (ms)**  
*Parameter group*

Signaling Type

**Access level**

Field

**Possible values**

0 through 6000

**Defaults**

1000

**Explanation**

Specifies how long WebSphere Voice Response waits before originating a new call after both the switch and WebSphere Voice Response have returned to idle (on-hook) following a previous call (incoming or outgoing) on that channel.

The Incoming Guard Time must be equal to, or less than, the Outgoing Guard Time.
Override SIP Transport IP Address

Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
A valid IP address

Defaults
Null

Explanation
The IP Address on which the SIP signaling stack runs if the multiple network connections are available on a single machine.
Page length for reports
Parameter group

General

Access level
Admin

Possible values
30 through 200

Defaults
60

Explanation
Specifies the page length for reports in lines.
Password Minimum Length
Parameter group
Application Server Interface

Access level
Admin

Possible values
2 through 8

Defaults
4

Explanation
Specifies the minimum length for all passwords that control access to a voice application mailbox.
Pause Key

Parameter group
Application Server Interface

Access level
Admin

Possible values
#
*
0 through 9

Defaults
8

Explanation
Specifies the key on the telephone keypad that can be used to pause, then restart during the playing of a voice segment, voice message, user greeting, or audio name. The default value means that callers must press 8 to pause, then press 8 again to restart.
Phone Number

Parameter group

Channel

Access level

Admin

Possible values

A 1- to 12-character string (0 through 9, A, B, C, and D)

Defaults

5551234

Explanation

Identifies the channel. In some conditions, depending on the value of the Call Information Type parameter for the Channel Group, this number, and the Area Code specified for the channel group, retrieve an application profile that starts the application that WebSphere Voice Response runs when a call arrives on this channel. (For more information about how the phone number is used, see “Channel identification” on page 37.)

This value can be any number that you select to identify a particular channel, up to 12 characters long. The valid characters are the digits are 0 through 9 and the letters A, B, C, and D.
Play Latency - Max Allowable (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

5 through 10000 milliseconds

Defaults

100 milliseconds

Explanation

Specifies the maximum play latency time that is allowed before WebSphere Voice Response generates white (information) message 25202.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Play Latency - Recovered (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

5 through 10000 milliseconds

Defaults

30 milliseconds

Explanation

Specifies the play latency time to which the system must return before WebSphere Voice Response generates green (cleared) message 25203. This follows a condition where the system has exceeded the maximum allowable play latency time.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Play Latency Time - Alert (ms)

Parameter group
Application Server Interface

Access level
Admin

Possible values
5 through 10000 milliseconds

Defaults
1500 milliseconds

Explanation
Every time the play latency time exceeds the value of this parameter, WebSphere Voice Response logs white (information) message 1400.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Play Skip (Seconds)

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 30

Defaults

7

Explanation

Specifies the number of seconds of the message that WebSphere Voice Response skips when a caller skips forward or backward through a voice message or a voice segment.
Printer Queue
Parameter group

General

Access level
Admin

Possible values
A print queue name

Defaults
lp0

Explanation
Specifies the AIX print queue to which WebSphere Voice Response sends information to be printed. The value of this parameter must be a printer definition that was created with SMIT.

The value is initially set to null, which automatically selects the default printer as defined to AIX. Any other value defines that printer as the default printer for WebSphere Voice Response only. If you enter an incorrect device name (that is, one not defined to the AIX on the pSeries computer on which WebSphere Voice Response is running), you cannot print.
Profile Retrieval Time - Alert (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

10 through 20000 milliseconds

Defaults

1500 milliseconds

Explanation

Every time the profile-retrieval time for non-cached application profiles exceeds the value of this parameter, WebSphere Voice Response logs white (information) message 1402.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Profile Retrieval Time - Max Allowable (ms)

Parameter group
Application Server Interface

Access level
Admin

Possible values
10 through 20000 milliseconds

Defaults
1000 milliseconds

Explanation
Specifies a maximum average time to retrieve non-cached application profiles. If the average time exceeds this value, white (information) message 25204 is logged.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Profile Retrieval Time - Recovered (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

10 through 20000 milliseconds

Defaults

800 milliseconds

Explanation

Specifies the average profile-retrieval time (for non-cached application profiles) to which the system must return before WebSphere Voice Response generates green (cleared) message 25205. This follows a condition where the system has exceeded the maximum allowable profile-retrieval time and logged message 25204.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see "CHP Performance Metrics - Expiry Time (mins)" on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
**Progress Indicator description value**

**Parameter group**

Trunk Interface

**Access level**

Admin

**Possible values**

An integer value in the range of 0 to 255 corresponding to the hex value to be transmitted (as the Progress Description value, byte 4) in the Progress Indicator IE sent with the ALERTING message for an inbound call.

**Defaults**

-1 (disables the sending of Progress Indicator)

**Explanation**

In some ISDN trunk voice applications there is a requirement to play voice to a caller prior to the call being completely established. An example might be to say “to avoid being charged for this call please hang up now” prior to sending a CONNECT (when charging begins). Some ISDN switches require a Progress Indicator Information Element (IE) with the initial ALERTING message in order to allocate a voice path ahead of receiving a CONNECT message.

Some ISDN switches do not allocate a voice path until after a call has been connected (not until after a CONNECT has been received and call charging has begun). The result is that voice played by an ISDN application cannot be heard until after an inbound call has been answered.

The system parameter Progress Indicator description value in the Trunk Interface group has been added for use with E1 Euro ISDN and E1 QSIG. This parameter has a default value of -1 but otherwise has an integer value in the range of 0 to 255 corresponding to the hex value to be transmitted (as the Progress Description value, byte 4) in the Progress Indicator IE sent with the ALERTING message for an inbound call. The default value of -1 disables the sending of Progress Indicator.

**Note:** The Progress Indicator description value system parameter is independent of the generic (field and lab only) Channel Group parameter Connect Voice Channel Before Answer. Both need to be set for voice to be heard before connect.
**Prompt Volume Ceiling Default (dBm)**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

0 through 50

**Defaults**

0

**Explanation**

Specifies the difference, in decibels (dBm), between the maximum permissible volume on the line and the maximum volume at which prompts are played. If you make the difference larger, prompts are played more quietly. This parameter is intended for use with background music. If you want to raise the volume of background music between prompts, specify a difference of 2 decibels for this parameter and a difference of 7 decibels for the Music Volume Ceiling Default (dBm) parameter. The state table can then use the ControlMusic action to raise and lower the volume of the music relative to the background maximum.

WebSphere Voice Response ensures that the values of Prompt Volume Ceiling Default (dBm) and Music Volume Ceiling Default (dBm) cannot together cause the volume on the line to exceed the maximum permissible volume.

This parameter can be overridden in an application by the System: Prompt : Volume ceiling system variable (SV223).
Proxy Address

Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
A character string

Defaults
Null

Explanation
The address of the proxy that is to be used if the Proxy Mode parameter is set to local proxy. The format of this address should be a SIP URI, for example: proxy@uk.ibm.com or proxy@9.20.2.200
Proxy Mode

Parameter group

VoIP SIP Signaling

Access level

Admin

Possible values

None
Local Proxy
Automatic Services DNSSRV
Proxy Routing Table

Defaults

None

Explanation

Specifies the method that will be used to route the SIP calls.

None specifies that WebSphere Voice Response is not to use a proxy. If a gateway is to be used, the address should be entered in the Default Destination URI field. If this field is left blank calls are sent directly to the endpoint.

Local Proxy specifies that WebSphere Voice Response is to route all SIP messages through a single Proxy, which is defined in the Proxy Address field.

Automatic Services DNSSRV specifies that the routing address is to be determined from a list of values returned by a DNSSRV lookup. This allows multiple proxies to be defined. Proxies can be assigned to specific services and be allocated a weighting to support load balancing.

Note: The Automatic Services DNSSRV method is not currently supported. If selected, the Manual Services Routing Table method will be used in its place.

Proxy Routing Table: The Manual Services Routing Table defined in /usr/lpp/dirTalk/db/sys_dir/srv.init (where an example srv.init file can be found) allows SRV request records to be manually entered into the srv.init file. The information in this file will then be used for routing by SIP signaling. If the request records contain weights and priorities, scheduling of services is maintained according to RFC 2782. The srv.init file can be
configured dynamically whilst VoIP is running. Any changes made will be reflected within approximately 60 seconds of the file being saved.
Proxy Port

Parameter group
VoIP SIP Signaling

Access level
Admin

Possible values
1024 - 65535

Defaults
5060

Explanation
IP port of a local proxy.
Real Time Delete Outbound Messages
Parameter group

General

Access level
Admin

Possible values
On
Off

Defaults
Off

Explanation
Allows you to specify that outgoing sent messages are deleted in real time, rather than being deleted overnight by DBCLNUP. This can be useful for heavily loaded systems, where overnight deletion can impose an additional overhead. This parameter does not affect the operation of DBCLNUP itself.
Real Time Migrate Voice Files
Parameter group

General

Access level

Admin

Possible values

On
Off

Defaults

Off

Explanation

Controls the migration of voice files (audio names, greetings, messages etc). The **On** value enables you to migrate your files in one of two ways; files can be migrated to intermediate formats or to the latest formats. The **Off** value migrates files to the original directory formats. The new format gives faster performance for access to voice files, especially on larger systems. If you are using Unified Messaging for WebSphere Voice Response, ensure that you have imported the latest level before enabling this parameter. In an SSI environment, before enabling this parameter on any client, ensure that all clients have been updated to the same level. If clients are on the same level it is safe to enable the parameter either on some, or on all clients.
Reconnect Call Feature Code
Parameter group

Signaling Type

Access level
Admin

Possible values
A character string

Defaults
H

Explanation
Specifies up to 16 characters that control signals sent to the switch to reconnect a call. It is important to correctly specify the code that the switch requires for the feature to operate. The characters are:

0 through 9, #, *, A, B, C, or D
   Send the digit or character.

H   Send a “hook flash” signal. Do not wait for dial tone.

G   Send a “ground key” signal. Do not wait for dial tone.

. (period)
   Wait for dial tone. The maximum time to wait is specified by the Pause parameter on the ReconnectCall action or, if its value is 0, by the Maximum Dial Tone Wait system parameter.

, (comma)
   Pause for the time that is specified by the Pause parameter on the ReconnectCall action.

\n   Pause for the time that is specified in n-tenths of a second (one digit only).

Example: Set this value to H\5*1 to reconnect a call on a switch that is configured to:
1. Accept a Hook Flash
2. Delay 0.5 seconds
3. Send +1.
When Reconnect Call Request Signal is set to Signaling, this parameter does not apply; in this condition, hook flash is the method used to request a reconnection.
**Reconnect Call Request Signal**

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

Feature Code
Hook Flash

**Defaults**

Hook Flash

**Explanation**

Specifies the method by which WebSphere Voice Response transmits to the switch a request to reconnect the call. For new applications, specify Feature Code. The Hook Flash value (called Signaling in previous releases) is retained only for compatibility with Version 1 Releases 1 and 2.

When Hook Flash is selected, the Reconnect Call Feature Code does not apply.

**Note:** The reconnect feature is not supported by all signaling protocols.
Record DTMF Level (dBm)

Parameter group

Trunk Interface

Access level

Admin

Possible values

-43 through -10

Defaults

-43

Explanation

Specifies the detection threshold for DTMF tones received by WebSphere Voice Response during record actions such as RecordVoiceMessage.
Record Voice Maximum (Seconds)

Parameter group

General

Access level

Admin

Possible values

30 through 3600

Defaults

120

Explanation

Specifies the maximum amount of time that is allowed for recording audio data (voice segments and messages).
Record Voice Maximum Pause (Seconds)

Parameter group

General

Access level

Admin

Possible values

0 through 60

Defaults

10

Explanation

Specifies how long the system waits when the Pause key is pressed while recording audio data. This parameter applies to all audio data (voice segments and messages).
**Record Voice Warning Time (Seconds)**

**Parameter group**

General

**Access level**

Admin

**Possible values**

0 through 60

**Defaults**

10

**Explanation**

Specifies how much time remains to record audio data. The system sends a beep to indicate that the amount of time allotted for recording audio data is about to expire. This parameter applies to all audio data (voice segments and messages).
Redial Limitation - Failed List Capacity

Parameter group

ISDN Signaling

Access level

Admin

Possible values

10 through 10000

Defaults

1000

Explanation

The Redial Limitation facility allows the system to prevent calls to an individual number after a certain number of failures.

A failed call list is maintained, when a call fails it is added to the list. This parameter specifies the length of the list.
Redial Limitation - Maximum Consecutive Failures

Parameter group
ISDN Signaling

Access level
Field

Possible values
1 through 50

Defaults
3

Explanation
The Redial Limitation facility allows the system to prevent calls to an individual number after a certain number of failures.

This parameter specifies the maximum number of consecutive failures that can occur during the time period specified by Redial Limitation - Timeout. After this number has been reached the system will prevent further calls to that extension.
Redial Limitation - Significant Digits
Parameter group

ISDN Signaling

Access level

Admin

Possible values

0 through 50

Defaults

0

Explanation

The Redial Limitation facility allows the system to prevent calls to an individual number after a certain number of failures.

This parameter specifies what proportion of the number should be used in the failed list table. This enables area codes to be stripped from the dialed number. The default value of 0 makes all digits significant.
Redial Limitation - Timeout
Parameter group
ISDN Signaling

Access level
Field

Possible values
0 through 600 seconds

Defaults
180 seconds

Explanation
The Redial Limitation facility allows the system to prevent calls to an individual number after a certain number of failures in a certain time period.

This parameter specifies the time period over which failed calls will be monitored. Once this period has elapsed then the system will remove the extension from the failed call list.
Re-Enable DTTA After Irrecoverable Error

Parameter group

General

Access level

Field

Possible values

Disabled
Enabled

Defaults

Enabled

Explanation

Specifies whether the DTTA is to be automatically re-enabled after an irrecoverable error. Normally, you should leave this parameter set to Enabled to ensure maximum availability of trunks on a production system. However, to help with some types of problem diagnosis, IBM Service might ask you to set this parameter to Disabled. When the parameter is set to Disabled, you must use the System Monitor or the ASCII console to re-enable the pack after a hardware failure.

The maximum number of times the DTTA is re-enabled in a one hour period is specified by the Maximum Retries for Pack/DTTA Reenabling parameter.
Re-Enable Trunk After Irrecoverable Error

Parameter group

Trunk Interface

Access level

Field

Possible values

Disabled
Enabled

Defaults

Enabled

Explanation

Specifies whether the telephony pack is to be automatically re-enabled after an irrecoverable error. Normally, you should leave this parameter set to Enabled to ensure maximum availability of trunks on a production system. However, to help with some types of problem diagnosis, IBM Service might ask you to set this parameter to Disabled. When the parameter is set to Disabled, you must stop, then restart WebSphere Voice Response to regain the use of the adapter after a hardware failure.

The maximum number of times the trunk is re-enabled in one hour is specified by the Maximum Retries for Pack/DTTA Reenabling parameter.
Register Addresses on Startup
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
Specifies whether or not the .ini files located in $SYS_DIR/voip are used to register SIP addresses at one or more registrars. If set to Yes, .ini files will be scanned and the register processes started up when a SIP trunk is enabled, and stopped when all SIP trunks are disabled. If set to No, no registrations will be made at all.
Register Default Timeout (Minutes)

Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
Integer in the range 1 through 10080

Defaults
60

Explanation
Specifies the default expiry length of a SIP registration. This can be overridden by the $SYS_DIR/voip/master.ini file, the line of the secondary ini files pointed to by master.ini, or both.
Register Default User Agent

Parameter group

VoIP SIP Signalling

Applicability

DTEA and DTNA

Access level

Admin

Possible values

A String

Defaults

“IBM Websphere Voice Response”

Explanation

Specifies the default User Agent (which represents an end system) for a SIP registration. This can be overridden by the $SYS_DIR/voip/master.ini file, or the line of the secondary ini files pointed to by master.ini.
Register Length

Parameter group

Signaling Type

Access level

Admin

Possible values

1 through 15

Defaults

5

Explanation

Specifies the number of incoming address digits when the Incoming Address Register Type parameter is set to Fixed Length.
Remote Play/Record CA Time Out (Seconds)

Parameter group
Application Server Interface

Access level
Admin

Possible values
1 through 30

Defaults
5

Explanation
This parameter specifies the time period inside which a custom server must close the voice channel after the end of recording or playing voice using the RecordVoiceToHost or PlayVoiceFromHost actions.
Remote Play/Record Max Data (KBytes)

Parameter group
Application Server Interface

Access level
Admin

Possible values
32 through 64

Defaults
48

Explanation
Specifies the maximum size of the voice data buffers that can be used for recording and playing voice data. This parameter is always rounded up to the nearest 4K boundary.
Remote Play/Record Min Data (KBytes)

Parameter group
Application Server Interface

Access level
Admin

Possible values
4 through 64

Defaults
24

Explanation
For recording, this parameter specifies the amount of voice data that must be recorded before the digital trunk adapter device driver can satisfy voice data reads from the custom server.

For PlayVoiceFromHost, this parameter specifies the amount of voice data buffer memory that must be used by the digital trunk adapter device driver before it can start playing the voice down the line. This is important both for the initial playing and any restart conditions, which might occur after a device underrun error.

Note: When CA_Play_Voice_Stream() or CA_Play_Voice_Elements() are used it is important that the voice data is written in integer multiples of 4000 bytes or the voice data buffers used will be sparsely populated with voice data.

For playing, this parameter is always rounded down to the nearest 4K boundary.

This parameter determines when the CA_Poll() subroutine returns on a recording channel. See CA_Poll() in WebSphere Voice Response for AIX: Custom Servers.
Reverse Key

Parameter group

Application Server Interface

Access level

Admin

Possible values

#  
*  
0 through 9

Defaults

7

Explanation

Specifies the key on the telephone keypad that can be used to skip backward through a voice segment, voice message, user greeting, or audio name. The number of seconds skipped is specified by the Play Skip system parameter. The default value means that callers should press 7 to skip backward.
RFC3264 Media on-hold method
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
ip=0.0.0.0 media-on-hold method
RFC3264 defined media-on-hold method
ip=0.0.0.0 and RFC3264 defined media-on-hold methods

Defaults
ip=0.0.0.0 media-on-hold method

Explanation
Specifies whether WebSphere Voice Response should use the RFC3264 defined method for media on-hold (rather than the older ip=0.0.0.0 method), or a combination of both methods.
Ringing Off Maximum (ms)
Parameter group

Signaling Type

Access level
Admin

Possible values
0 through 18000

Defaults
5000

Explanation
Specifies the maximum amount of time WebSphere Voice Response expects between ringing-on signals from the switch. WebSphere Voice Response recognizes a valid ringing signal only when the signal matches the template that is defined by both the Ringing On and both the Ringing Off parameters.

This parameter is used only with those signaling protocols that use ringing to indicate incoming seizure: FXS Ground Start, FXS Loop Start, SAS Loop Start, RE (Remote Extension), SL (Subscriber Loop) and U.K. Exchange.
Ringing Off Minimum (ms)
Parameter group

Signaling Type

Access level
Admin

Possible values
0 through 5100

Defaults
400

Explanation
Specifies the minimum amount of time WebSphere Voice Response expects between ringing-on signals from the switch. WebSphere Voice Response recognizes a valid ringing signal only when the signal matches the template that is defined by both the Ringing On and both the Ringing Off parameters.

This parameter is used only with those signaling protocols that use ringing to indicate incoming seizure: FXS Ground Start, FXS Loop Start, SAS Loop Start, RE (Remote Extension), SL (Subscriber Loop) and U.K. Exchange.
Ringing On Maximum (ms)
Parameter group

Signaling Type

Access level
Admin

Possible values
0 through 5100

Defaults
2400

Explanation
Specifies the maximum amount of time WebSphere Voice Response expects for a ringing-on signal from the switch. WebSphere Voice Response recognizes a valid ringing signal only when the signal matches the template that is defined by both the Ringing On and both the Ringing Off parameters.

This parameter is used only with those signaling protocols that use ringing to indicate incoming seizure: FXS Ground Start, FXS Loop Start, SAS Loop Start, RE (Remote Extension), SL (Subscriber Loop) and U.K. Exchange.
Ringing On Minimum (ms)

Parameter group

Signaling Type

Access level
Admin

Possible values
0 through 5100

Defaults
600

Explanation

Specifies the minimum amount of time WebSphere Voice Response expects for a ringing-on signal from the switch. WebSphere Voice Response recognizes a valid ringing signal only when the signal matches the template that is defined by both the Ringing On and both the Ringing Off parameters.

This parameter is used only with those signaling protocols that use ringing to indicate incoming seizure: FXS Ground Start, FXS Loop Start, SAS Loop Start, RE (Remote Extension), SL (Subscriber Loop) and U.K. Exchange.
RTCP Enable Sender Report
Parameter group
VoIP DTEA and DTNA Media

**Applicability**
DTEA and DTNA

**Access level**
Admin

**Possible values**
No
Yes

**Defaults**
No

**Explanation**
Defines whether to enable the sending of RTCP reports.
RTCP Sender Report Interval
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA and DTNA

Access level
Admin

Possible values
1 through 60.

Defaults
5

Explanation
The interval at which RTCP reports will be sent (if RTCP Enable Sender Report is enabled).
Override DTNA RTP Transport IP Address

Parameter group
VoIP DTEA and DTNA Media

Applicability
DTNA only

Access level
Admin

Possible values
Null, or a ‘dotted’ IP address such as 9.20.38.97

Defaults
Null

Explanation
This parameter controls the IP address used for DTNA Media (RTP). If left at the default setting (Null) or set to Null, WebSphere Voice Response selects the first valid IP address on the system unit. If set to a ‘dotted’ IP address, WebSphere Voice Response uses this address for sending and receiving RTP Media. The specified IP address must exist as a network address on the system unit otherwise WebSphere Voice Response will issue an error message when the VoIP channels are enabled.
RTP Base Port Number

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA and DTNA

Access level

Admin

Possible values

2 through 65534

Defaults

6000

Explanation

The base port number to be used by the adapter to assign to a media channel. The adapter will use this value as the base for assigning 480 ports for RTP and RTCP. The base port number should always be an even number.

Note: For DTNA, ports can be calculated as follows:

For trunk T (1...16) and Channel C (1...24 or 1...30)
- RTP port = ((T-1)\times30 + C-1)\times2 + Base Port
- RTCP port = ((T-1)\times30 + C-1)\times2 + Base Port + 1
RTP IP TOS Byte (TOS)
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTEA and DTNA

Access level
Admin

Possible values
0 through 63

Defaults
0

Explanation
The Type of Service (TOS) setting for RTP packets being sent across the network. A higher setting will give RTP packets priority over standard IP packets.
RTP IP Time to Live (TTL)

Parameter group

VoIP DTEA and DTNA Media

Applicability

DTEA and DTNA

Access level

Admin

Possible values

1 through 255

Defaults

20

Explanation

Time to Live (TTL) for RTP packets sent over the network.
RTP Security Negotiation
Parameter group
VoIP DTEA and DTNA Media

Applicability
DTNA

Access level
Admin

Possible values
Unsecured
Secure
Both

Defaults
Unsecured

Explanation
Secure RTP provides confidentiality and message authentication to RTP data. It can be used to prevent people from listening to or tampering with the audio data sent over a unsecure network like the internet. In WebSphere Voice Response, there are three possible RTP security configuration settings:

Unsecured
The default. WebSphere Voice Response does not accept secure RTP for inbound or offer secure RTP for outbound calls. Inbound calls that only offer secure RTP will be rejected with a 488 Not Accepted Here response.

Secure
WebSphere Voice Response only uses secure RTP. Inbound calls not capable of secure RTP are rejected with a 488 Not Accepted Here response, and outbound calls made by WebSphere Voice Response will only offer secure RTP.

Both
For inbound calls WebSphere Voice Response accepts secure RTP if offered, but will also accept calls if only RTP is offered. In the case of both secure RTP and RTP being offered, secure RTP will be used. For outbound calls, both secure RTP and RTP will be offered.
Partial support is provided for optional crypto session parameters [RFC 4568 section 6.3]. This support is enabled when Secure RTP has been configured for either Secure or Both.

The following session parameters are fully supported:

- UNENCRYPTED_SRTP
- UNENCRYPTED_SRTCP
- UNAUTHENTICATED_SRTP

All other session parameters are parsed, but are not supported. Any crypto lines containing the unsupported parameters are ignored, and treated as unsuitable matches. If there are no other suitable matches (which can be either unsecure RTP/AVP, or crypto attributes with supported session parameters) the SDP is rejected. This will result in a SIP response of 488 Not Acceptable Here.

Session parameters will never be presented on outbound SIP requests/responses. This includes outbound INVITE (make call or on hold requests) and responses to OPTIONS.

For more information, refer to “Secure RTP” in the WebSphere Voice Response: Voice over IP using Session Initiation Protocol manual.
**Runtime Cache Check Interval (Seconds)**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

5 through 600 seconds

**Defaults**

60 seconds

**Explanation**

In a single system image, this parameter specifies the time after which WebSphere Voice Response tests whether any application objects have been changed.

When WebSphere Voice Response first uses an application object (such as a state table), it stores that object in a shared memory cache so that it does not have to reload the object from disk next time the object is used. When the properties of an object are changed, the runtime cache is updated:

- On a stand-alone WebSphere Voice Response system, the runtime cache is usually updated immediately, so the new properties become effective immediately. However, if you change the object properties by importing data from a tape or disk, the runtime cache is not updated until the runtime cache check interval ends.

- In a WebSphere Voice Response system that is configured as a single system image, the runtime cache is updated immediately on the system on which the changes were made. However, the caches on the other nodes in the cluster are not updated until the runtime cache check interval expires.

To decide how to set this parameter, consider the following:

- A low value for this parameter means that the runtime cache is updated frequently. This means the cache is more likely to be up-to-date when the application object is used, but the frequent caching requires more transactions with the application object database.

- A high value for this parameter means that the runtime cache is updated less frequently, so fewer transactions affect the application object database.
However, the cached data persists for longer, so it is more likely to be out-of-date when the application object is used.
SDI Inter-trunk staggering delay (s)

Parameter group

General

Access level

Field

Possible values

0 through 600 seconds

Defaults

0 seconds

Explanation

This parameter causes a delay following an attempt to disable a trunk before any other enables occur. This might be used to stagger enabling of trunks where the sudden loading causes system or application problems. The default is 0, which means no additional delay.
SDI Timeout - Channel Disable

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

15 seconds

Explanation

Specifies the maximum time in which a signaling process must respond to an
SL_CHANNEL_DISABLE_REQ primitive by sending an
SL_CHANNEL_DISABLE_CNF primitive. After this time, WebSphere Voice
Response disables the channel without waiting any longer.

Do not change this value unless you understand exactly what you are doing.
Otherwise you might badly affect the operation of the system. If you have
problems, return to the default value.
SDI Timeout - Channel Enable
Parameter group

General

Access level
Field

Possible values
0 through 3600 seconds

Defaults
250 seconds

Explanation
Specifies the maximum time in which a signaling process must respond to an
SL_CHANNEL_ENABLE_REQ primitive by sending an
SL_CHANNEL_ENABLE_CNF primitive. After this time, WebSphere Voice
Response ends the attempt to enable the channel, leaving it disabled.

Do not change this value unless you understand exactly what you are doing.
Otherwise you might badly affect the operation of the system. If you have
problems, return to the default value.
SDI Timeout - Channel Outservice
Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

60 seconds

Explanation

Specifies the maximum time in which a signaling process must respond to an
SL_CHANNEL_DISABLE_REQ primitive by sending an
SL_CHANNEL_DISABLE_CNF primitive. After this time, WebSphere Voice
Response disables the channel without waiting any longer. This time-out
value differs from the SDI Timeout - Channel Disable, because it applies only
if a trunk is disabled when the channels are in use.

Do not change this value unless you understand exactly what you are doing.
Otherwise you might badly affect the operation of the system. If you have
problems, return to the default value.
SDI Timeout - Pack Diagnostics

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

500 seconds

Explanation

Specifies the maximum time WebSphere Voice Response is to wait for diagnostics on a pack to complete. After this time, WebSphere Voice Response ends the attempt to enable the pack.

Do not change this value unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
SDI Timeout - Reco Statistics Reset

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

30 seconds

Explanation

Specifies the maximum time WebSphere Voice Response is to wait for speech recognition statistics to be read.

Do not change this value unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
SDI Timeout - Signaling Process Reconfiguration

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

60 seconds

Explanation

Specifies the maximum time WebSphere Voice Response is to wait for all registered signaling processes to respond to an SL_RECONFIG_TRUNK_REQ primitive with an SL_RECONFIG_TRUNK_CNФ primitive. After this time, WebSphere Voice Response continues enabling the trunk, but it is possible that out-of-date configuration data might be in use.

On a heavily loaded system with many registered signaling processes, it might be necessary to increase this value, but do not change it unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
SDI Timeout - SL_TRUNK_DISABLE_REQ

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

120 seconds

Explanation

Specifies the maximum time in which a signaling process must respond to an SL_TRUNK_DISABLE_REQ primitive by sending an SL_TRUNK_DISABLE_CNF primitive. After this time, WebSphere Voice Response disables the trunk without waiting any longer.

Do not change this value unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
SDI Timeout - SL_TRUNK_ENABLE_REQ

Parameter group

General

Access level

Field

Possible values

0 through 3600 seconds

Defaults

120 seconds

Explanation

Specifies the maximum time in which a signaling process must respond to an SL_TRUNK_ENABLE_REQ primitive by sending an SL_TRUNK_ENABLE_CNF primitive. After this time, WebSphere Voice Response ends the attempt to enable the trunk, leaving it disabled.

Do not change this value unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
**SDI Timeout - Trunk Disable**

**Parameter group**

General

**Access level**

Field

**Possible values**

0 through 3600 seconds

**Defaults**

60 seconds

**Explanation**

Specifies the maximum time WebSphere Voice Response is to wait for all channels on a trunk to be disabled. After this time, WebSphere Voice Response disables all trunks without waiting any longer.

Do not change this value unless you understand exactly what you are doing. Otherwise you might badly affect the operation of the system. If you have problems, return to the default value.
Secure SIP Enabled
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
False
True

Defaults
False

Explanation
This controls whether or not WebSphere Voice Response can accept secure inbound calls and can make secure outbound calls. Setting the Secure SIP Enabled parameter alone will not allow WebSphere Voice Response to make secure SIP calls. You must have previously set up the keyring.db certificate database. For instructions on how to do this, refer to “Secure SIP minimal configuration” in the WebSphere Voice Response: Voice over IP using Session Initiation Protocol manual.
Seize Acknowledgment Timeout (ms)

Parameter group

Signaling Type

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 5100

Defaults

200

Explanation

Specifies the maximum time allowed for the switch to acknowledge an outgoing seizure by WebSphere Voice Response. This parameter applies only to the R2 signaling protocol.
### Send RAI

**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

No  
Yes

**Defaults**

No

**Explanation**

Specifies whether WebSphere Voice Response is to send a T1 remote alarm indication (RAI), also known as a *yellow alarm*, to the switch while the trunk operating status is *Enabled* and the channels are out of service. The RAI is discontinued when WebSphere Voice Response completes a channels *In Service* action for all channels on the trunk. This is used with the Blocking Action parameter to block channels from receiving incoming calls.
Send RESTART on Channel Enable

Parameter group

ISDN Signaling

Access level

Admin

Possible values

No
Yes

Defaults

Yes

Explanation

Specifies whether WebSphere Voice Response is to send a RESTART message when channels are enabled.
Session Timer Allow Update For Refresh
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
This controls whether reINVITE or UPDATE are to be used for refresh (assuming that far-end has said that it will allow the use of UPDATE using the ALLOW header).
**Session Timer Enable**

**Parameter group**
VoIP SIP Signalling

**Applicability**
DTEA and DTNA

**Access level**
Admin

**Possible values**
Disabled
Inbound only
Outbound only
Enabled

**Defaults**
Disabled

**Explanation**
Allows the Session Timer to be disabled completely, enabled for inbound calls only, enabled for outbound calls only, or enabled for both directions.
Session Timer Inbound Refresher Default

Parameter group

VoIP SIP Signalling

Applicability

DTEA and DTNA

Access level

Admin

Possible values

UAC (Remote Endpoint) - the call originator
UAS (WebSphere Voice Response = Local Endpoint)

Defaults

UAC (Remote End Point)

Explanation

Specifies whether, on an inbound call (WebSphere Voice Response acting as UAS=Uase Agent Server) and when no refresher is specified, on the Session Timer offer, whether the UAS (WebSphere Voice Response) or the UAC (Remote End Point) will be selected as the session timer refresher.
Session Timer Maximum Session Time

Parameter group

VoIP SIP Signalling

Applicability

DTEA and DTNA

Access level

Admin

Possible values

90 through 86400 (24 hours)

Defaults

1800 (30 minutes)

Explanation

Maximum allowed session time in seconds.
Session Timer Minimum Session Time
Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
90 through 86400 (24 hours)

Defaults
90

Explanation
Minimum allowed session time in seconds. This controls the value put in the 'Min-SE' header of a Session Timer offer.
**Session Timer Outbound Calls Refresher Default**

**Parameter group**

VoIP SIP Signalling

**Applicability**

DTEA and DTNA

**Access level**

Admin

**Possible values**

None Requested  
UAC (WebSphere Voice Response = Local Endpoint) Requested  
UAS (Remote Endpoint) Requested

**Defaults**

None Requested

**Explanation**

Specifies whether, on an outbound call (WebSphere Voice Response acting as UAC=User Agent Client) the choice of refresher will be none (left to remote endpoint to make decision), UAC (WebSphere Voice Response will be refresher) or UAS (forces far end to be refresher).
Settle Time (ms)

Parameter group

Trunk Interface

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 60000

Defaults

15000

Explanation

Specifies the amount of time WebSphere Voice Response allows for the network to settle whenever the system is restarted.
**Signaling Process Type**

**Parameter group**

Channel Group

**Access level**

Admin

**Possible values**

ACL
ISDN - 5ESS - 5E8
ISDN - 5ESS - 5E9
ISDN - TR41449
ISDN - DMS100 - BCS34
ISDN - Euro-ISDN
ISDN - E1 QSIG
ISDN DMS 100 National 2
ISDN T1 National 2
ISDN Japanese INS
None
SMSI/SMDI/VMS
User1 through User19

**Defaults**

None

**Explanation**

Specifies the signaling process type to be used for an exchange data link or for common channel signaling.

Select **ISDN - 5ESS - 5E8** or **ISDN - 5ESS - 5E9** (depending on the level of software you have on the switch), when the Call Information Type parameter is set to Signaling Process and you are using the AT&T/Lucent 5ESS ISDN protocol.

Select **ISDN - TR41449**, when the Call Information Type parameter is set to Signaling Process and you are using the AT&T/Lucent Definity ISDN TR41449 or TR41459 protocol.

Select **ISDN - DMS100 - BCS34** when the Call Information Type parameter is set to Signaling Process and you are using the Northern Telecom DMS100 ISDN protocol.
Select **ISDN - Euro-ISDN** when the Call Information Type parameter is set to Signaling Process and you are using the Euro-ISDN protocol.

Select **ISDN - E1 QSIG** when the Call Information Type parameter is set to Signaling Process and you are using the QSIG protocol.

**ISDN DMS 100 National 2** is a valid value, but it is not yet supported.

Select **ISDN T1 National 2** when the Call Information Type parameter is set to Signaling Process and you are using the ISDN National 2 protocol on the Summa Four switch, or the 5E12 protocol on the Lucent 5ESS-2000 switch.

Select **ISDN - Japanese INS** when the Call Information Type parameter is set to Signaling Process and you are using the INS Net Service 1500 protocol.

Select **SMSI/SMDI/VMS** or **ACL** when the Call Information Type parameter is set to Signaling Process and you are using the exchange data link (EDL). You must also select the appropriate EDL Switch Type (in the Exchange Data Link parameter group).

Select **User1** through **User19** only if you provide your own signaling process type (see *WebSphere Voice Response for AIX: Programming for the Signaling Interface*).
Signaling Trunk Identifier

Parameter group

Trunk Interface

Access level

Admin

Possible values

A character string

Defaults

(None)

Explanation

Identifies the trunk for the purposes of message information that is sent on an exchange data link or a signaling channel when common channel signaling is in use. The way the value is used depends on the signaling process.

If you are using ISDN, set the ISDN Trunk Identifier parameter, instead of this parameter.

If you are using SMSI, this value must be the character string that identifies the trunk.
**Signaling Type**

**Parameter group**

Channel Group

**Access level**

Admin

**Possible values**

- 1 through 16
- DID
- E&M
- EL7/CAS (Ericsson MD110)
- FXS Ground Start
- FXS Loop Start
- Italy
- R2
- RE
- ROLM E&M
- ROLM FXS Loop Start
- SAS Loop Start
- SL
- TS003
- UK CallStream
- UK Exchange
- UK Tie/DDI

**Explanation**

Specifies the signaling type definition to be used by the channels in the channel group. The signaling types (1 through 16 and the named types listed under Possible Values) specify values for several signaling parameters (see “Signaling type parameter group” on page 189 for more information).

Signaling Type is not used for common channel signaling protocols (including ISDN).
SNA Status Refresh Period (seconds)

Parameter group

Application Server Interface

Access level

Admin

Possible values

2 through 600

Defaults

10

Explanation

 Specifies how frequently the status of 3270 sessions is refreshed in the 3270 Monitor window.
SSI Custom Server Status Check Interval (seconds)

Parameter group
Application Server Interface

Access level
Admin

Possible values
5 through 32767 seconds

Defaults
60 seconds

Explanation
In a single system image, this parameter specifies the time after which WebSphere Voice Response tests whether any custom servers have been installed or uninstalled. After this time, WebSphere Voice Response updates the custom server information that is shown in the Custom Server Manager window and the ASCII console, and it updates the information that is returned to applications through SNMP.

To decide how to set this parameter, think about the following:
- A low value for this parameter means that the information is updated frequently. This means the information is more likely to be up-to-date, but the frequent testing requires more transactions with the application object database.
- A high value for this parameter means fewer transactions affect the application object database, but the custom server status data persists for longer, so it is more likely to be out-of-date when it is used.

On a stand-alone WebSphere Voice Response system, this parameter has no effect.
Start Java and VoiceXML Environment Automatically

Parameter group

General

Access level
Admin

Possible values
No
Yes

Defaults
Yes

Explanation
The Yes value specifies whether the Java and VoiceXML environment is to be started and ended at the same time as Voice Response is started and shutdown. Set the value to No if you are running a network of Java and VoiceXML nodes on multiple machines (a plex). If you set the value to No, you must use line commands to start and end the HostManagers and nodes. See WebSphere Voice Response for AIX: Deploying and Managing VoiceXML and Java Applications for more information.
State Table Entry Label

Parameter group
Application Server Interface

Access level
Admin

Possible values
A state table entry label

Defaults
Start

Explanation
Specifies the entry point of the state table that answers incoming calls. If you write your own state table to answer incoming calls, you might need to reset this parameter in addition to the State Table Name for Incoming Calls parameter.
State Table Loop Detection

Parameter group

Application Server Interface

Access level

Admin

Possible values

Disabled
Enabled

Defaults

Disabled

Explanation

Specifies whether loop detection is activated.
State Table Loop Detection Loop Analysis Threshold
Parameter group
Application Server Interface

Access level
Admin

Possible values
0 through 65535

Defaults
50

Explanation
Specifies the number of times a state table can return to the same state before loop analysis begins. When the state table has returned to the same state for four times this value (for example, 4 × 50 = 200 times), alarm 506 is logged and the succeeding states are recorded in a log file in the $CUR_DIR/oamlog directory. When the number of states recorded is twice the value of the State Table Loop Detection: Maximum Length parameter (for example, 2 × 30 = 60), alarm 507 is logged and the state table is automatically ended.
State Table Loop Detection Loop Threshold

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 65535

Defaults

20

Explanation

Specifies the number of times a loop can be executed after loop analysis begins, before the state table is ended automatically.
State Table Loop Detection Maximum Length

Parameter group

Application Server Interface

Access level

Admin

Possible values

0 through 65535

Defaults

30

Explanation

Specifies the maximum number of states that can be assumed to be a loop. At least twice this number of states must be traced before a loop can be detected reliably. The log file might contain entries for many more states.
**State Table Name for Incoming Calls**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

A state table name

**Defaults**

Incoming_Call

**Explanation**

Specifies the state table that answers all incoming calls. If you write your own state table for this purpose or rename Incoming_Call, reset this parameter.
Stop Key

Parameter group

Application Server Interface

Access level

Admin

Possible values

#

* 0 through 9

Defaults

*

Explanation

Specifies the key on the telephone keypad that can be used to stop the playing of a voice segment, voice message, user greeting, or audio name. The default value means that callers should press * to stop.
Subnet Mask

Parameter group

VoIP Media - Adapters

Access level

Admin

Possible values

0.0.0.0 through 255.255.255.255

Defaults

255.255.255.0

Explanation

Records the IPv4 IP address subnet mask of a DTEA card. There are four instances of this parameter, one for each card.
**Switch Encoding Law**

**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

Default
A law
μ law

**Defaults**

Default

**Explanation**

Specifies how voice data is to be encoded on the connection to the switch. This is normally μ-law if the trunk interface is T1, or A-law if the trunk interface is E1 (as specified by the Trunk Interface parameter in the WebSphere Voice Response parameter group). Select A law if you want to use this algorithm with a T1 trunk interface or select μ law if you want to use this algorithm with an E1 interface. Otherwise, leave it as Default.
Switch Type

Parameter group

Trunk Interface

Access level

Admin

Possible values

- Aspect Call Center
- AT&T/Lucent 4ESS
- AT&T/Lucent 5ESS-2000
- AT&T/Lucent 5ESS/4ESS
- AT&T System 75/85
- AT&T/Lucent Definity G1/G2/G3
- Australian PTT
- DMS 100/MTX
- BT
- Centrex
- Channel Bank
- Default
- Ericsson MD110
- Finland PTT
- France Telecom
- Germany PTT
- GPT iSDX
- Harris 20-20
- China (Hong Kong S.A.R.) PTT
- Intecom IBX
- Italy PSTN CAS
- Mercury (UK)
- NEC NEAX 61E
- NT DMS100/250
- NT Meridian (SL1)
- Other
- Portugal PTT
- Rockwell Galaxy
- ROLM 9751 (9005)
- ROLM/Siemens 9751 (9006)
- Siemens Hicom 300
- Spain PTT
- Summa Four
- Swiss PTT
- Taiwan PTT

Defaults

Default

Explanation

Specifies the type of switch, channel bank, service, or service provider to which WebSphere Voice Response is connected. This parameter is used only by Pack Configuration to help determine suitable values for other telephony configuration parameters.
System Default Application Profile
Parameter group

General

Access level
Admin

Possible values
An application profile ID

Defaults
0000000000

Explanation
Specifies the default Application Profile ID. Before starting the Incoming_Call state table, WebSphere Voice Response looks for an application profile ID that matches the called number, if the called number is available. If the called number is not available, WebSphere Voice Response looks for an application profile ID that matches the channel identification (Area Code and Phone number). If no match is found, the system looks for an application profile that matches the value specified by this parameter.

If no matching application profile exists, WebSphere Voice Response does not answer the call.
System Disk Threshold
Parameter group

General

Access level
Admin

Possible values
0 through 100 percent

Defaults
90 percent

Explanation
Specifies the amount of space in the WebSphere Voice Response file systems that can be used before WebSphere Voice Response displays a warning on the System Monitor Console to indicate that the system is running out of disk space. In addition, when this amount of space has been used, the CheckStorage state table action returns the result “Resource Unavailable” for disk storage.
System Language

Parameter group

General

Access level

Field

Possible values

1 through 255

Defaults

1

Explanation

The language that is used for WebSphere Voice Response window text. The default is U.S. English.
**System Monitor Graph Duration (Minutes)**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

1 through 65535

**Defaults**

20

**Explanation**

 Specifies the duration of the graphs in the System Monitor window. The period that you specify is rounded up to the nearest 4-minute duration.
**System Name**

**Parameter group**

General

**Access level**

Admin

**Possible values**

A character string

**Defaults**

Not applicable

**Explanation**

Specifies the name of the system that is to be used on reports. If this is not specified, the hostname of the pSeries computer is used.
System Number

Parameter group

Exchange Data Link

Access level

Admin

Possible values

A character string

Defaults

01

Explanation

The message system for which the message waiting indicator is intended. This parameter is used only by VMS exchange data links.
System Response during Server Outage

Parameter group
Application Server Interface

Access level
Admin

Possible values
Play technical difficulties message
Busy-out all telephony channels
Do not answer calls (calls will continue to ring)

Explanation

If a Data Server outage occurs, either of File or DB2, this parameter controls how the system responds to calls. The following options are supported:

Play technical difficulties message
All calls in progress are immediately terminated, and the system technical difficulties message will be played for all new incoming calls.

Busy-out all telephony channel
All calls in progress are immediately terminated, and all telephony channels are disabled. Once the data servers are again available the telephony channels in service when the outage occurred will be restored to service.

Do not answer calls
Incoming callers will continue hearing ringing tone. Voice Response will not answer any calls during the outage.
T1 Bit Robbing

Parameter group

Trunk Interface

Access level

Field

Possible values

On
Off

Defaults

On

Explanation

Specifies whether robbed-bit signaling is to be used on T1 trunks. Robbed-bit signaling causes a signal-to-distortion ratio approximately 1.8 dB lower than that for 8-bit coding. With extended superframe format (ESF) on ISDN trunks, robbed bit signaling is not used, so select Off if you are using a T1 pack for ISDN. “Clear channel” (that is, 64 Kbit per second) operation is possible only if bit robbing is not used. This parameter does not apply to E1 trunks.
T1 CAS Protocol

Parameter group

Channel Group

Access level

Admin

Possible values

E&M
FXS
DID
SAS

Defaults

E&M

Explanation

Specifies the T1 channel associated signaling (CAS) protocol that is used by channels in this group. This parameter is not used with common channel signaling (CCS).

Note: Select E&M instead of DID when the Call Information Type parameter is set to Register. DirectTalk DID is the same signaling protocol as E&M. The DID value is retained only for compatibility with previous releases.
**T1 CAS Signaling Format**

**Parameter group**

Signaling Type

**Access level**

Admin

**Possible values**

2-bit AB (SF)
4-bit ABCD (ESF)

**Defaults**

2-bit AB (SF)

**Explanation**

Specifies the T1 bit-robbing CAS Signaling format (as defined in TIA/EIA-464-B) that is to be used. If the T1 framing mode on the trunk is set to D4, choose 2-bit format. If the T1 framing mode on the trunk is ESF, choose 4 bit.

2-bit format means that only the AB bits are used for signaling; C and D are ignored.

4-bit format means that all the ABCD bits are used for signaling. The C and D bits are normally set to the values of the A and B bits respectively, but this depends on the protocol (for more information, see TIA/EIA-464-B).

Because the signaling format is defined on a channel group basis, the user must ensure that channel groups that span multiple trunks all have the trunks set to use either D4 or ESF; otherwise some signaling problems might occur.
T1 Framing Mode

Parameter group

Trunk Interface

Access level

Admin

Possible values

D3/D4
ESF

Defaults

D3/D4

Explanation

Specifies the frame format that is to be used with T1 trunks. Select D3/D4 for superframe format. Select ESF for extended superframe format.

Superframe format timeshares the framing bit to identify channel framing and signaling channel framing. Channel framing identifies the location of time slot 1 and signaling framing identifies those frames in which signaling bits a and b are transmitted using robbed-bit signaling.

The extended superframe format extends the superframe structure from 12 frames to 24 frames and redefines the 8kbits per second framing bit position. The 8kbits per second ESF channel is divided into 2kbits per second for channel framing and signaling channel framing, 2kbits per second for Cyclic Redundancy Check code (CRC-6) and 4kbits per second for a data link. Four signaling channels, A, B, C, and D, are provided by ESF.

ESF is set automatically when a T1 common channel signaling protocol (such as ISDN) is selected. When a pack is used with a channel associated signaling protocol, you must select D3/D4 or ESF as necessary.

See “Setting line code and framing mode parameters” on page 119 for guidance on which value to select.
**T1 Hit Filter (1.5 ms)**

**Parameter group**

Trunk Interface

**Access level**

Field

**Possible values**

6 through 20 units

**Defaults**

7 units (10.5 ms)

**Explanation**

Specifies how long a signaling state change must exist to qualify as valid. This value is adjustable in units of 1.5 ms. The default value sets this parameter to 10.5 ms.

To define this parameter, enter a number that is two-thirds of the desirable hit filter time. For example, to define a hit filter of 13.5 ms, enter 9.

WebSphere Voice Response uses this parameter only when the trunk interface is defined as T1.
**T1 Line Code**

**Parameter group**

Trunk Interface

**Access level**

Admin

**Possible values**

AMI  
B8ZS

**Defaults**

AMI

**Explanation**

Specifies which T1 line coding scheme WebSphere Voice Response uses to ensure a ones density that is enough for proper timing recovery of the T1 digital signal.

When alternate mark inversion (AMI) is selected, the WebSphere Voice Response line coding scheme is AMI with zero code suppression (ZCS). With AMI, binary one bits are represented by alternate positive and negative pulses (marks) and binary zero bits are represented by spaces. AMI with ZCS changes the second least significant bit (bit 7) from a zero bit (space) to a one bit (pulse) whenever 8 successive zeros occur. The bit is not restored to a 0 at the receiving end.

**Note:** Some equipment manufacturers refer to AMI with ZCS as *bit 7 stuffing*. (AMI without ZCS is often referred to as *transparent*.)

When Bipolar Eight Zero Substitution (B8ZS) is selected, the WebSphere Voice Response line coding scheme is AMI with B8ZS. B8ZS uses bipolar violations to ensure a good enough ones density. Each block of eight successive zeros is replaced by the B8ZS code 000VB0VB, where “B” represents an inserted pulse that conforms to the AMI rule and “V” represents an AMI violation (also known as a bipolar violation (BPV)). At the receiving end, the B8ZS code is replaced by 8 zeros.

B8ZS is required to provide clear channel capability, which allows data rates to 64 Kbits per second. Clear channel operation is possible only with common channel signaling such as ISDN.
**Note:** Both the switch and WebSphere Voice Response must be using the same T1 line coding scheme or errors occur.

See “Setting parameters for hangup tone detection” on page 112 for guidance on which value to select.
T1 Remote Alarm Format

Parameter group

Trunk Interface

Access level

Field

Possible values

RAI via bit 2 = 0 in every channel
RAI via FS bit of frame 12

Defaults

RAI via bit 2 = 0 in every channel

Explanation

Specifies which T1 remote alarm format to use.
T.38 Fax Refer URI

Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
URI String, for example, 10.20.31.42

Defaults
Null

Explanation
Specifies the URI to which a SIP REFER will be sent when T.38 fax is detected in an incoming INVITE.
**Time in Cache (minutes)**

**Parameter group**
Application Server Interface

**Access level**
Admin

**Possible values**
1 through 20

**Defaults**
5

**Explanation**
Specifies how long the system holds a voice segment in cache memory when it has a request for buffers and no other buffers are available. This parameter defines the minimum amount of time that must elapse between when a voice segment now in cache memory was last requested and when WebSphere Voice Response releases the buffer. WebSphere Voice Response does not release a buffer containing a voice segment until it has “timed out” as specified by this parameter.

This parameter *must* be set to a time that is longer than the longest voice segment that is to be played using a prompt. If it is set to a shorter time, voice segments that are still playing might be released.
Time Off 1 Maximum (0.001 Seconds)

Parameter group

Call Progress Tones

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 50000

Defaults

0

Explanation

With Time Off 1 Minimum, Time On 1 Minimum, and Time On 1 Maximum, defines how long the first frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time Off 1 Minimum (0.001 Seconds)**

**Parameter group**
Call Progress Tones

**Access level**
Field

**Possible values**
Integer multiples of 20, in the range 0 through 50000

**Defaults**
0

**Explanation**
With Time Off 1 Maximum, Time On 1 Minimum, and Time On 1 Maximum, defines how long the first frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time Off 2 Maximum (0.001 Seconds)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Integer multiples of 20, in the range 0 through 50000

**Defaults**

0

**Explanation**

With Time Off 2 Minimum, Time On 2 Minimum, and Time On 2 Maximum, defines how long the second frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time Off 2 Minimum (0.001 Seconds)**

**Parameter group**
Call Progress Tones

**Access level**
Field

**Possible values**
Integer multiples of 20, in the range 0 through 50000

**Defaults**
0

**Explanation**
With Time Off 2 Maximum, Time On 2 Minimum, and Time On 2 Maximum, defines how long the second frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time Off 3 Maximum (0.001 Seconds)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Integer multiples of 20, in the range 0 through 50000

**Defaults**

0

**Explanation**

With Time Off 3 Minimum, Time On 3 Minimum, and Time On 3 Maximum, defines how long the third frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
Parameter group

Call Progress Tones

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 50000

Defaults

0

Explanation

With Time Off 3 Maximum, Time On 3 Minimum, and Time On 3 Maximum, defines how long the third frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time On 1 Maximum (0.001 Seconds)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Integer multiples of 20, in the range 0 through 50000

**Defaults**

0

**Explanation**

With Time On 1 Minimum, Time Off 1 Minimum, and Time Off 1 Maximum, defines how long the first frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Time On 1 Minimum (0.001 Seconds)**

**Parameter group**
Call Progress Tones

**Access level**
Field

**Possible values**
Integer multiples of 20, in the range 0 through 50000

**Defaults**
0

**Explanation**
With Time On 1 Maximum, Time Off 1 Minimum, and Time Off 1 Maximum, defines how long the first frequency component of the call progress tone continues. For more information, see "Setting call progress tone parameters for outbound dialing" on page 101.
**Time On 2 Maximum (0.001 Seconds)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Integer multiples of 20, in the range 0 through 50000

**Defaults**

0

**Explanation**

With Time On 2 Minimum, Time Off 2 Minimum, and Time Off 2 Maximum, defines how long the second frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
Time On 2 Minimum (0.001 Seconds)

Parameter group
Call Progress Tones

Access level
Field

Possible values
Integer multiples of 20, in the range 0 through 50000

Defaults
0

Explanation
With Time On 2 Maximum, Time Off 2 Minimum, and Time Off 2 Maximum, defines how long the second frequency component of the call progress tone continues. For more information, see "Setting call progress tone parameters for outbound dialing" on page 101.
**Time On 3 Maximum (0.001 Seconds)**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Integer multiples of 20, in the range 0 through 50000

**Defaults**

0

**Explanation**

With Time On 3 Minimum, Time Off 3 Minimum, and Time Off 3 Maximum, defines how long the third frequency component of the call progress tone continues. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
Time On 3 Minimum (0.001 Seconds)

Parameter group

Call Progress Tones

Access level

Field

Possible values

Integer multiples of 20, in the range 0 through 50000

Defaults

0

Explanation

With Time On 3 Maximum, Time Off 3 Minimum, and Time Off 3 Maximum, defines how long the third frequency component of the call progress tone continues. For more information, see "Setting call progress tone parameters for outbound dialing" on page 101.
Tone Group

Parameter group

Channel Group

Access level

Field

Possible values

Tone Group 1 through 12

Defaults

Tone Group 1

Explanation

Specifies the call progress tone group for the channels in this channel group. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.
**Tone Label**

**Parameter group**
Call Progress Tones

**Access level**
Field

**Possible values**
A character string

**Defaults**
Unused

**Explanation**

Specifies the specific call progress tone, in the basic tone type. For more information, see [“How call progress tones are defined” on page 105](#).
**Tone Type**

**Parameter group**

Call Progress Tones

**Access level**

Field

**Possible values**

Busy
Dial
Network Busy
Ring
Other

**Defaults**

Unused

**Explanation**

Specifies the basic type to which the call progress tone belongs. For more information, see "How call progress tones are defined" on page 105.
Transfer Call Feature Code
Parameter group

Signaling Type

Access level
Admin

Possible values
A character string

Defaults
H

Explanation
Specifies up to 16 characters that control signals sent to the switch to transfer a call. It is important to correctly specify the code that is required by the switch for the feature to operate. The characters are:

0 through 9, #, *, A, B, C, or D
    Send the digit or character.
H    Send a “hook flash” signal. Do not wait for dial tone.
G    Send a “ground key” signal. Do not wait for dial tone.
. (period)
    Wait for dial tone. The maximum time to wait is specified by the Pause parameter on the TransferCall action or, if its value is 0, by the Maximum Dial Tone Wait system parameter.
, (comma)
    Pause for the time that is specified by the Pause parameter on the TransferCall action.
\n    Pause for the time that is specified in n-tenths of a second (one digit only).

Example: Set this value to H\5+7\5 to transfer a call on a switch that is configured to:
1. Accept a Hook Flash
2. Delay 0.5 seconds
3. Send +7
4. Delay another 0.5 seconds before sending the telephone number.
**Note:** Set this value to \(H\backslash5\times7\). for a switch that is configured to send dial tone after the *7.

When Transfer Call Request Signal is set to Signaling, this parameter does not apply; in this condition, hook flash is the method that is used to request a transfer.
Transfer Call Request Signal

Parameter group

Signaling Type

Access level
Admin

Possible values
Feature Code
Hook Flash

Defaults
Hook Flash

Explanation
Specifies the method by which WebSphere Voice Response transmits to the switch a request to transfer the call. For new applications, specify Feature Code. The Hook Flash value (called Signaling in previous releases) is kept only for compatibility with Version 1 Releases 1 and 2.

When Hook Flash is selected, the Transfer Call Feature Code does not apply.

Note: The transfer feature is not supported by all signaling protocols.
Transport Protocol

Parameter group

VoIP SIP Signaling

Access level

Admin

Possible values

UDP
TCP
TLS

Defaults

UDP

Explanation

The protocol to be used for sending all signaling messages for calls initiated by WebSphere Voice Response. Inbound calls are to be accepted from both TCP and UDP protocols.
**Trunk Interface**

**Parameter group**

General

**Access level**

Admin

**Possible values**

E1/A-law
T1/µ-law

**Defaults**

T1/µ-law

**Explanation**

Specifies the type of interface that is between the trunk and WebSphere Voice Response. The choice of interface type also affects the voice encoding scheme that is used. A-law and µ-law are the voice encoding schemes normally used by E1 and T1 interfaces respectively. However, it is possible to use A-law encoding with a T1 interface; to do so, set the Switch Encoding Law parameter to A-law.
Trunk Interlock - 3270 Server
Parameter group

General

Access level

Admin

Possible values

Enabled
Disabled

Defaults

Enabled

Explanation

The Enabled value specifies that, if 3270 is installed, WebSphere Voice Response is to wait until all 3270 sessions that are configured with 3270 servers are ready before it automatically initializes the trunks. If you have 3270 installed but no 3270 sessions configured, select Disabled.

Note: This parameter was called Trunk Initialization Interlock - 3270 Server in previous releases.
Trunk Interlock - Java and VoiceXML Environment

Parameter group

General

Access level

Admin

Possible values

Enabled
Disabled

Defaults

Enabled

Explanation

The Enabled value specifies that the Java and VoiceXML environment must be initialized before Voice Response enables channels to start taking calls. If this parameter is disabled, channels are enabled without waiting for the Java and VoiceXML environment to initialize.

Note: This parameter is relevant only if Start Java and VoiceXML Environment Automatically is set to yes.
Trunk Interlock EDL
Parameter group

Trunk Interface

Access level

Admin

Possible values

Disabled
Enabled

Defaults

Enabled

Explanation

The Enabled value specifies that, if a trunk is associated with an exchange data link, the exchange data link process is essential for normal operation. Whenever an attempt is made to enable the trunk (including at startup), WebSphere Voice Response waits for the time that is specified by the Trunk Interlock EDL Timeout parameter and, if the exchange data link process is still inactive, the trunk is not enabled.

In addition, if the exchange data link process stops while the trunk is active, WebSphere Voice Response disables the trunk.

The Disabled value specifies that the exchange data link process is not essential. WebSphere Voice Response enables the trunk at startup, and keeps the trunk enabled, whether the exchange data link process is running or not.
Trunk Interlock EDL Timeout (minutes)

Parameter group

General

Access level

Admin

Possible values

1 through 30 (minutes)

Defaults

3 (minutes)

Explanation

Specifies how long WebSphere Voice Response waits for an exchange data link process to become active before it automatically enables the trunk. This parameter is used only when the following conditions are true:

- The trunk has been configured to enable automatically (for example, Operating Status is set to Inservice)
- The trunk has an exchange data link that is configured for one or more of its channel groups
- Trunk Interlock - EDL is set to Disabled.
Trunk Interlock Inservice Delay (seconds)

**Parameter group**
General

**Access level**
Admin

**Possible values**
0 through 600

**Defaults**
0

**Explanation**
Specifies the delay between each telephony trunk that comes into service during WebSphere Voice Response initialization. In a system with multiple trunks, this “gentle startup” avoids the problem that occurs when all channels on all trunks come into service at the same time and start to answer calls, when the required state tables, prompts, and voice segments are not already cached in memory.
Trunk Interlock Timeout (minutes)
Parameter group
General

Access level
Admin

Possible values
1 through 30

Defaults
10

Explanation
Specifies the timeout period for automatic trunk initialization. If the preconditions have not been met during this period, the attempt to enable any further trunks is stopped and a red alarm is raised.

Note: This parameter was called Trunk Initialization Timeout in previous releases.
Trunk Signaling Mode

Parameter group

Trunk Interface

Access level

Field
CAS
CCS-SP
ISDN

Defaults

CAS

Explanation

Specifies the type of signaling protocol that is to be used on the trunk. Select CAS if you are using channel associated signaling. Select CCS-SP if you are using a custom-written common channel signaling process. Select ISDN if you are using one of the IBM-supplied ISDN features.
UK Tie/DDI Start Type
Parameter group

Channel Group

Access level
Admin

Possible values
Immediate Start
Delay Start

Defaults
Delay Start

Explanation
Specifies the start signal that WebSphere Voice Response uses for all channels in the channel group, both for incoming and outgoing address signaling.

With delay start operation, address digits are not sent until after the start signal (off-hook/on-hook) is received by the switch. Immediate start allows address signals to be sent immediately (for example, inside 65 ms) after seizure by the switch.

If the Call Information Type parameter for the Channel Group is set to Register, it is best to use only delay start operation. With immediate start, it is possible that address digits might be missed if a register is not attached before digits are sent. However, immediate start operation is best if Call Information Type is set to Signaling Process or None.

Note: If the switch requires “inverted E&M” (that is, European E&M) and is using wink start, it is acceptable to select Delay Start, provided that the default values are used for Delay Start Delay and Delay Start Duration.
**Underrun Margin Time - Alert (ms)**

**Parameter group**
Application Server Interface

**Access level**
Admin

**Possible values**
50 through 60000 milliseconds

**Defaults**
500 milliseconds

**Explanation**
Every time the underrun margin time falls below the value of this parameter, WebSphere Voice Response logs white (information) message 1401.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Underrun Margin Time - Min Allowable (ms)

Parameter group

Application Server Interface

Access level

Admin

Possible values

50 through 60000 milliseconds

Defaults

1000 milliseconds

Explanation

Specifies the minimum underrun margin time that is allowed before WebSphere Voice Response generates alarm message 25200.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.
Underrun Margin Time - Recovered (ms)

Parameter group
Application Server Interface

Access level
Admin

Possible values
50 through 60000 milliseconds

Defaults
2000 milliseconds

Explanation
Specifies the underrun margin time to which the system must return before WebSphere Voice Response generates green (cleared) message 25201. This follows a condition where the system has fallen below the minimum allowable underrun margin time.

The message is generated only if the CHP Performance Metrics - Expiry Time (mins) system parameter (see “CHP Performance Metrics - Expiry Time (mins)” on page 247) is set to a value greater than 0.

For more information, see “Monitoring the performance of a single system image” on page 150.

Use allowed host list

Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin
Possible values

No
Yes

Defaults

No

Explanation

Specifies whether or not to use an allowed host list to exclude SIP Requests from unlisted IP addresses.
Use SIP REQHDR for Application Profile Selection

Parameter group
VoIP SIP Signalling

Applicability
DTEA and DTNA

Access level
Admin

Possible values
No
Yes

Defaults
No

Explanation
Specifies which of TO or Request headers is used for called number (and Application Profile) selection.
**User Greeting Compression Type**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

Compressed
Uncompressed

**Defaults**

Compressed

**Explanation**

Specifies whether user greetings are recorded compressed or uncompressed, when using the RecordUserGreeting state table action.

It is best to set this parameter when you install WebSphere Voice Response, and not to change it afterward. This prevents your having some user greetings that are compressed and some that are uncompressed on the same system; WebSphere Voice Response can then search more effectively.

An uncompressed user greeting occupies 5 times more disk space than the same user greeting does if it is compressed.

If much interference occurs when the user greeting is recorded, you might find that compressing the user greeting results in a poor quality of playback.

*If you want to change this parameter from its default setting, ensure that all your custom servers that use the CA_Get_Greeting_Info subroutine can process a returned GREETING_INFO_ST structure in which the compression_type field can have any valid value. If you do not, the data returned will be corrupted.*

Restart your WebSphere Voice Response system after changing this parameter.
**User Identifier Minimum Digits**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

1 through 10

**Defaults**

3

**Explanation**

Defines the number of characters for which WebSphere Voice Response waits before it begins to search for a matching string when a state table does the GetFindName action.
**Voice Interrupt Detection Level (dBm)**

**Parameter group**

Channel Group

**Access level**

Admin

**Possible values**

-47 through 0

**Defaults**

-43

**Explanation**

Specifies the level that the audio signal must reach to trigger voice interrupt detection. When the level of the audio signal exceeds and remains above this level, WebSphere Voice Response interprets it as a voice interrupt.

For a voice interrupt to be detected by WebSphere Voice Response, three criteria must be met:

- The energy level of the audio signal must be equal to, or exceed, the value that is specified by the Voice Interrupt Detection Level parameter.
- The energy level of the audio signal must remain equal to, or above, this level for the time that is specified by the Voice Interrupt Detection On Time parameter.
- The energy level of the audio signal must then remain below the value that is specified by the Voice Interrupt Detection On Time parameter.

The value of this system parameter is used as the initial value for the `System : Voice interrupt detection : Level` system variable (SV218) for any application that uses this channel group. An application can override this initial setting. For information about this system variable see the *WebSphere Voice Response for AIX: Application Development using State Tables* book.

See also the Voice Interrupt Detection On Time parameter and the Voice Interrupt Detection Off Time parameter.
Voice Interrupt Detection Off Time (ms)

Parameter group

Channel Group

Access level

Admin

Possible values

0 through 1000

Defaults

200

Explanation

Specifies the time the audio signal must stay below the Voice Interrupt Detection Level after the interrupt has qualified using the value in the Voice Interrupt Detection On Time parameter.

Values are rounded down to the nearest 20 ms.

The value of this system parameter is used as the initial value for the System : Voice interrupt detection : Off time system variable (SV220) for any application that uses this channel group. An application can override this initial setting. For information about this system variable see the WebSphere Voice Response for AIX: Application Development using State Tables book.

See also the Voice Interrupt Detection Level parameter and the Voice Interrupt Detection On Time parameter.
Voice Interrupt Detection On Time (ms)

Parameter group

Channel Group

Access level

Admin

Possible values

100 through 200

Defaults

160

Specifies the time the level of the audio signal must stay above the Voice Interrupt Detection Level before WebSphere Voice Response interprets it as a voice interrupt.

Values are rounded down to the nearest 20 ms.

The value of this system parameter is used as the initial value for the system variable System: Voice interrupt detection: On time(SV219) for any application that uses this channel group. An application can override this initial setting. For information about this system variable see the WebSphere Voice Response for AIX: Application Development using State Tables book.

See also the Voice Interrupt Detection Level parameter and the Voice Interrupt Detection Off Time parameter.
Voice Message Compression Type
Parameter group
Application Server Interface

Access level
Admin

Possible values
Compressed
Uncompressed

Defaults
Compressed

Explanation
Specifies whether voice messages are recorded in a compressed or uncompressed format, when using the RecordVoiceMessage state table action.

You should set this parameter when you install WebSphere Voice Response, and then not change it afterwards. This prevents you having both compressed and uncompressed voice messages on the same system—WebSphere Voice Response can then search more effectively.

Note that an uncompressed voice message occupies 5 times more disk space than it would do when compressed.

If the messages you are recording are of reduced quality, for example because they originate from a mobile phone or compressed voice-over-IP network, then using uncompressed rather than compressed message storage within WebSphere Voice Response can improve the playback quality.

If you want to change this parameter from its default setting, ensure that all your custom servers that use the CA_Get_Voice_Msg_Info subroutine can process a returned VOICE_MSG_INFO_ST structure in which the compression_type field can have any valid value. If you do not, the data returned will become corrupted.

Restart your WebSphere Voice Response system after changing this parameter.
**Voice Message ID Prefetch**

**Parameter group**

Application Server Interface

**Access level**

Admin

**Possible values**

1 through 4096

**Defaults**

128

**Explanation**

Each time a voice message is recorded, a new voice message identifier is required. If some identifiers are already allocated, a new database transaction is not required. This parameter specifies how many voice message identifiers WebSphere Voice Response allocates at one time.

If your WebSphere Voice Response system is configured as a single system image, many client telephony channels probably exist, each one producing voice messages. In this type of environment, performance is improved if this parameter is set to a high value.
Voice Table Index (Characters)

Parameter group

Application Server Interface

Access level

Admin

Possible values

A character string

Defaults

ABCDEFGHIJKLMNOPQRSTUVWXYZ

Specifies the entries in the voice table that are specified by Voice Table Name (Characters) in the sequence that they appear in the table. The index is used by the CHARACTERS prompt statement to determine which voice segment to play.

For an individual state table, this parameter can be overridden by the System : Voice table index : Characters system variable.
Voice Table Name (Characters)

Parameter group

Application Server Interface

Access level

Admin

Possible values

A voice table name

Defaults

Alphabet

Explanation

Specifies the name of the voice table that catalogs the voice segments that speak the letters of the alphabet. Voice applications use this table to speak letters (using the CHARACTERS prompt statement, for example). WebSphere Voice Response is delivered with a voice table named Alphabet that contains all the letters. If you rename this table or catalog the letters into a different table, be sure to reset this parameter.

For an individual state table, this parameter can be overridden by the System : Voice name : Characters system variable.
Voice Table Name (Digits)

Parameter group

Application Server Interface

Access level

Admin

Possible values

A voice table name

Defaults

Numbers

Explanation

Specifies the name of the voice table that catalogs the voice segments that speak digits. Voice applications use this table to speak numbers (using the DIGITS prompt statement, for example). WebSphere Voice Response is delivered with a voice table named Numbers that contains all the digits. If you rename this table or catalog the digits into a different table, be sure to reset this parameter.

For an individual state table, this parameter can be overridden by the System : Voice table name : Digits system variable.
Wink Start Delay (ms)
Parameter group

Signaling Type

Access level

Field

Possible values

100 through 5000

Defaults

120

Explanation

Specifies how much time elapses after WebSphere Voice Response receives the off hook signal from the switch but before it sends the wink start signal to the switch. This parameter is important only when channels are using wink start as the start type. The start type is determined by one of the Start Type parameters in the Channel Group parameter group.
Wink Start Duration (ms)
Parameter group

Signaling Type

Access level

Field

Possible values
120 through 280

Defaults
200

Explanation

Specifies the length of the wink start signal that WebSphere Voice Response sends to the switch. This parameter is important only when channels are using wink start as the start type. The start type is determined by one of the Start Type parameters in the Channel Group parameter group. Also specifies the minimum length of the wink start signal WebSphere Voice Response expects from the switch when WebSphere Voice Response originates a call.
Appendix B. System parameter templates

This appendix lists the values set in the supplied templates:

- “Signaling type templates”
- “Trunk interface templates” on page 585
- “Call progress tone templates” on page 595

Related Information

- “Using system parameter templates” on page 17
- “Defining trunk interfaces” on page 75
- “Defining signaling types” on page 78
- “Setting call progress tone parameters for outbound dialing” on page 101

Signaling type templates

Templates are provided for the signaling protocols that are listed here.

This section describes the values of the WebSphere Voice Response system parameters as they are supplied in each template. Because so many templates are available, they are described in four tables; to find the table that you want, see Table 20.

Note: In the tables, a blank cell indicates that the parameter has no meaning for the protocol to which the template applies.

<table>
<thead>
<tr>
<th>Signaling Protocol</th>
<th>Parameters Are Listed in</th>
</tr>
</thead>
<tbody>
<tr>
<td>Aspect E&amp;M</td>
<td>Table 21 on page 576</td>
</tr>
<tr>
<td>DID (T1)</td>
<td>Table 21 on page 576</td>
</tr>
<tr>
<td>E&amp;M - including ROLM 9751 (9006)</td>
<td>Table 21 on page 576</td>
</tr>
<tr>
<td>EL7/CAS (Ericsson MD110) (E1)</td>
<td>Table 23 on page 581</td>
</tr>
<tr>
<td>FXS and SAS Loop Start</td>
<td>Table 22 on page 578</td>
</tr>
</tbody>
</table>

5. Signaling Type does not apply to common channel signaling protocols.
Table 20. Templates for signaling protocols (continued)

<table>
<thead>
<tr>
<th>Signaling Protocol</th>
<th>Parameters Are Listed in</th>
</tr>
</thead>
<tbody>
<tr>
<td>FXS Ground Start</td>
<td>Table 21</td>
</tr>
<tr>
<td>Italy (E1)</td>
<td>Table 22 on page 578</td>
</tr>
<tr>
<td>R2 Digital Line Signaling (E1)</td>
<td>Table 24 on page 583</td>
</tr>
<tr>
<td>RE (Remote Extension) (E1)</td>
<td>Table 22 on page 578</td>
</tr>
<tr>
<td>ROLM 9751 (9005) E&amp;M</td>
<td>Table 21</td>
</tr>
<tr>
<td>ROLM 9751 (9005) FXS Loop Start</td>
<td>Table 22 on page 578</td>
</tr>
<tr>
<td>SL (Subscriber Loop) (E1)</td>
<td>Table 23 on page 581</td>
</tr>
<tr>
<td>TS003/P2 Line Signaling (E1)</td>
<td>Table 24 on page 583</td>
</tr>
<tr>
<td>U.K. CallStream (E1)</td>
<td>Table 23 on page 581</td>
</tr>
<tr>
<td>U.K. Exchange (E1)</td>
<td>Table 23 on page 581</td>
</tr>
<tr>
<td>U.K. Tie/DDI (E1)</td>
<td>Table 23 on page 581</td>
</tr>
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</table>

Table 21. Signaling types: template values (1)

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<thead>
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<th>System Parameter</th>
<th>Access Level</th>
<th>DID (T1)</th>
<th>E&amp;M - including ROLM 9751 (9006)</th>
<th>ROLM 9751 (9005) E&amp;M</th>
<th>Aspect E&amp;M</th>
<th>FXS Ground Start</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer Delay Time</td>
<td>Admin</td>
<td>260 ms</td>
<td>260 ms</td>
<td>260 ms</td>
<td>260 ms</td>
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</tr>
<tr>
<td>Blocking Action</td>
<td>Field</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Other</td>
<td>Other</td>
</tr>
<tr>
<td>Cadence Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
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<tr>
<td>Cadence Energy Minimum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>Cadence Off Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence Off Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence On Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
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</table>
Table 21. Signaling types: template values (1) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>DID (T1)</th>
<th>E&amp;M - including ROLM 9751 (9006)</th>
<th>ROLM 9751 (9005) E&amp;M</th>
<th>Aspect E&amp;M</th>
<th>FXS Ground Start</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cadence On Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence Silence Maximum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
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<tr>
<td>CO Acknowledgment</td>
<td>Field</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
</tr>
<tr>
<td>CO Off-Hook</td>
<td>Field</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
</tr>
<tr>
<td>CO On-Hook</td>
<td>Field</td>
<td>700 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td>40 ms</td>
</tr>
<tr>
<td>Constant Energy Minimum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
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<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
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<tr>
<td>Delay Start Delay</td>
<td>Field</td>
<td>120 ms</td>
<td>120 ms</td>
<td>120 ms</td>
<td>120 ms</td>
<td>120 ms</td>
</tr>
<tr>
<td>Delay Start Duration</td>
<td>Field</td>
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<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
</tr>
<tr>
<td>DialPause</td>
<td>Admin</td>
<td>200 ms</td>
<td>200 ms</td>
<td>600 ms</td>
<td>200 ms</td>
<td>200 ms</td>
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<tr>
<td>Dial Tone Qualify Time</td>
<td>Admin</td>
<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td>200 ms</td>
</tr>
<tr>
<td>Dial Tone (Detection) Timeout</td>
<td>Admin</td>
<td>8000 ms</td>
<td>8000 ms</td>
<td>8000 ms</td>
<td>8000 ms</td>
<td>8000 ms</td>
</tr>
<tr>
<td>Glare Detection Time</td>
<td>Field</td>
<td>40 ms</td>
<td>40 ms</td>
<td>40 ms</td>
<td>40 ms</td>
<td>40 ms</td>
</tr>
<tr>
<td>Ground Flash</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hangup Detection</td>
<td>Admin</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
</tr>
<tr>
<td>Hook Flash</td>
<td>Admin</td>
<td>300 ms</td>
<td>300 ms</td>
<td>500 ms</td>
<td>500 ms</td>
<td></td>
</tr>
<tr>
<td>Incoming Address Register Type</td>
<td>Admin</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td></td>
</tr>
<tr>
<td>Incoming Guard Time</td>
<td>Field</td>
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<td>0 ms</td>
<td>0 ms</td>
<td>0 ms</td>
<td>40 ms</td>
</tr>
<tr>
<td>No Answer Warning</td>
<td>Admin</td>
<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
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<tr>
<td>Outgoing Address Register Type</td>
<td>Admin</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
<td></td>
</tr>
<tr>
<td>Outgoing Guard Time</td>
<td>Field</td>
<td>1000 ms</td>
<td>1000 ms</td>
<td>1000 ms</td>
<td>1000 ms</td>
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</table>

Appendix B. System parameter templates 577
Table 21. Signaling types: template values (1) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>DID (T1)</th>
<th>E&amp;M - including ROLM 9751 (9006)</th>
<th>ROLM 9751 (9005) E&amp;M</th>
<th>Aspect E&amp;M</th>
<th>FXS Ground Start</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reconnect Call Feature Code</td>
<td>Admin</td>
<td>H</td>
<td>H(5+1) H</td>
<td>H</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Reconnect Call Request Signal</td>
<td>Admin</td>
<td>Hook Flash</td>
<td>Feature Code</td>
<td>Hook Flash</td>
<td>Hook Flash</td>
<td></td>
</tr>
<tr>
<td>Register Length</td>
<td>Admin</td>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Ringing Off Maximum</td>
<td>Admin</td>
<td></td>
<td>5000 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing Off Minimum</td>
<td>Admin</td>
<td></td>
<td>400 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Maximum</td>
<td>Admin</td>
<td></td>
<td>2400ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Minimum</td>
<td>Admin</td>
<td></td>
<td>600 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Seize Acknowledgment Timeout</td>
<td>Field</td>
<td></td>
<td>200 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>T1 CAS Signaling Format</td>
<td>Admin</td>
<td>2-bit AB (SF)</td>
<td>2-bit AB (SF)</td>
<td>2-bit AB (SF)</td>
<td>2-bit AB (SF)</td>
<td>2-bit AB (SF)</td>
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<tr>
<td>Transfer Call Feature Code</td>
<td>Admin</td>
<td>H</td>
<td>H(5+7) H</td>
<td>H</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transfer Call Request Signal</td>
<td>Admin</td>
<td>Hook Flash</td>
<td>Feature Code</td>
<td>Feature Code</td>
<td>Hook Flash</td>
<td></td>
</tr>
<tr>
<td>Wink Start Delay</td>
<td>Field</td>
<td>120 ms</td>
<td>120 ms</td>
<td>120 ms</td>
<td>120 ms</td>
<td></td>
</tr>
<tr>
<td>Wink Start Duration</td>
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<td>200 ms</td>
<td>200 ms</td>
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</table>

Table 22. Signaling types: template values (2)

<table>
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<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>FXS and SAS Loop Start</th>
<th>ROLM 9751 (9005) FXS Loop Start</th>
<th>Italy (E1)</th>
<th>RE (Remote Extension) (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer Delay Time</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Blocking Action</td>
<td>Field</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
</tr>
<tr>
<td>Cadence Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
</tr>
</tbody>
</table>
### Table 22. Signaling types: template values (2) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>FXS and SAS Loop Start</th>
<th>ROLM 9751 (9005) FXS Loop Start</th>
<th>Italy (E1)</th>
<th>RE (Remote Extension) (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cadence Energy Minimum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>Cadence Off Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence Off Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence On Time Minimum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence On Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence Silence Maximum</td>
<td>Admin</td>
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<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>CO Acknowledgment</td>
<td>Field</td>
<td>60 ms</td>
<td>60 ms</td>
<td>20 ms</td>
<td>60 ms</td>
</tr>
<tr>
<td>CO Off-Hook</td>
<td>Field</td>
<td></td>
<td></td>
<td>40 ms</td>
<td></td>
</tr>
<tr>
<td>CO On-Hook</td>
<td>Field</td>
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<td></td>
<td>160 ms</td>
<td></td>
</tr>
<tr>
<td>Constant Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
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<td>Constant Energy Minimum</td>
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<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
</tr>
<tr>
<td>Delay Start Delay</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Delay Start Duration</td>
<td>Field</td>
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<tr>
<td>Dial Pause</td>
<td>Admin</td>
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<tr>
<td>Dial Tone Qualify Time</td>
<td>Admin</td>
<td>200 ms</td>
<td>200 ms</td>
<td></td>
<td>200 ms</td>
</tr>
<tr>
<td>Dial Tone (Detection) Timeout</td>
<td>Admin</td>
<td>8000 ms</td>
<td>8000 ms</td>
<td></td>
<td>8000 ms</td>
</tr>
<tr>
<td>Glare Detection Time</td>
<td>Field</td>
<td>40 ms</td>
<td>40 ms</td>
<td>40 ms</td>
<td>40 ms</td>
</tr>
<tr>
<td>Ground Flash</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td>120 ms</td>
</tr>
<tr>
<td>Hangup Detection</td>
<td>Admin</td>
<td>Constant Energy Detection</td>
<td>Constant Energy Detection</td>
<td>Off</td>
<td>Constant Energy Detection</td>
</tr>
<tr>
<td>Hook Flash</td>
<td>Admin</td>
<td>500 ms</td>
<td>500 ms</td>
<td></td>
<td>100 ms</td>
</tr>
<tr>
<td>System Parameter</td>
<td>Access Level</td>
<td>FXS and SAS Loop Start</td>
<td>ROLM 9751 (9005) FXS Loop Start</td>
<td>Italy (E1)</td>
<td>RE (Remote Extension) (E1)</td>
</tr>
<tr>
<td>----------------------------------------</td>
<td>--------------</td>
<td>------------------------</td>
<td>----------------------------------</td>
<td>------------</td>
<td>----------------------------</td>
</tr>
<tr>
<td>Incoming Address Register Type</td>
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<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Incoming Guard Time</td>
<td>Field</td>
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<td>860 ms</td>
<td>100 ms</td>
<td>860 ms</td>
</tr>
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<td>No Answer Warning</td>
<td>Admin</td>
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<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
</tr>
<tr>
<td>Outgoing Address Register Type</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Outgoing Guard Time</td>
<td>Field</td>
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<td>1500 ms</td>
<td>100 ms</td>
<td>1500 ms</td>
</tr>
<tr>
<td>Reconnect Call Feature Code</td>
<td>Admin</td>
<td>H</td>
<td>H\5*1</td>
<td>H</td>
<td></td>
</tr>
<tr>
<td>Reconnect Call Request Signal</td>
<td>Admin</td>
<td>Hook Flash</td>
<td>Feature Code</td>
<td>Hook Flash</td>
<td></td>
</tr>
<tr>
<td>Register Length</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing Off Maximum</td>
<td>Admin</td>
<td>5000 ms</td>
<td>5000 ms</td>
<td>5000 ms</td>
<td></td>
</tr>
<tr>
<td>Ringing Off Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
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<td>Ringing On Maximum</td>
<td>Admin</td>
<td>2400 ms</td>
<td>2400 ms</td>
<td>1260 ms</td>
<td></td>
</tr>
<tr>
<td>Ringing On Minimum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>220 ms</td>
<td></td>
</tr>
<tr>
<td>Seize Acknowledgment Timeout</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
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<td>TI CAS Signaling Format</td>
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<td>2-bit AB (SF)</td>
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<td></td>
</tr>
<tr>
<td>Transfer Call Feature Code</td>
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<td>H</td>
<td>H\5*7\5</td>
<td>H</td>
<td></td>
</tr>
<tr>
<td>Transfer Call Request Signal</td>
<td>Admin</td>
<td>Hook Flash</td>
<td>Feature Code</td>
<td>Hook Flash</td>
<td></td>
</tr>
<tr>
<td>Wink Start Delay</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wink Start Duration</td>
<td>Field</td>
<td></td>
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<td></td>
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</tr>
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Table 23. Signaling types: template values (3)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>SL (Subscriber Loop) (E1)</th>
<th>U.K. CallStream (E1)</th>
<th>U.K. Exchange (E1)</th>
<th>U.K. Tie/DDI (E1)</th>
<th>EL7/CAS (Ericsson MD110) (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer Delay Time</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>260 ms</td>
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<tr>
<td>Blocking Action</td>
<td>Field</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
<td>Offhook</td>
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<tr>
<td>Cadence Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
</tr>
<tr>
<td>Cadence Energy Minimum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>Cadence Off Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence Off Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
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</tr>
<tr>
<td>Cadence On Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
<td>600 ms</td>
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</tr>
<tr>
<td>Cadence On Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence Silence Maximum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>CO Acknowledgment</td>
<td>Field</td>
<td>60 ms</td>
<td>20 ms</td>
<td>20 ms</td>
<td>20 ms</td>
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<tr>
<td>CO Off-Hook</td>
<td>Field</td>
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<td>0 ms</td>
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<td>CO On-Hook</td>
<td>Field</td>
<td>200 ms</td>
<td>160 ms</td>
<td>160 ms</td>
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<td></td>
</tr>
<tr>
<td>Constant Energy Maximum</td>
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<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
<td>0 dBm</td>
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<tr>
<td>Constant Energy Minimum</td>
<td>Admin</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
</tr>
<tr>
<td>Delay Start Delay</td>
<td>Field</td>
<td></td>
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<td></td>
<td></td>
<td>20 ms</td>
</tr>
<tr>
<td>Delay Start Duration</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>160 ms</td>
</tr>
<tr>
<td>Dial Pause</td>
<td>Admin</td>
<td>2700 ms</td>
<td>200 ms</td>
<td>200 ms</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Dial Tone Qualify Time</td>
<td>Admin</td>
<td>1000 ms</td>
<td>200 ms</td>
<td>200 ms</td>
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<td></td>
</tr>
<tr>
<td>Dial Tone (Detection) Timeout</td>
<td>Admin</td>
<td>12000 ms</td>
<td>60000 ms</td>
<td>80000 ms</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Glare Detection Time</td>
<td>Field</td>
<td>40 ms</td>
<td>40 ms</td>
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<td>Access Level</td>
<td>SL (Subscriber Loop) (E1)</td>
<td>U.K. CallStream (E1)</td>
<td>U.K. Exchange (E1)</td>
<td>U.K. Tie/DDI (E1)</td>
<td>EL7/CAS (Ericsson MD110) (E1)</td>
</tr>
<tr>
<td>-------------------------------</td>
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<td>-------------------</td>
<td>-----------------</td>
<td>-------------------------------</td>
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<td>Ground Flash</td>
<td>Field</td>
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<tr>
<td>Hangup Detection</td>
<td>Admin</td>
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</tr>
<tr>
<td>Hook Flash</td>
<td>Admin</td>
<td>100 ms</td>
<td>100 ms</td>
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<td></td>
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<td>Incoming Address Register Type</td>
<td>Admin</td>
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<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Incoming Guard Time</td>
<td>Field</td>
<td>500 ms</td>
<td>100 ms</td>
<td>100 ms</td>
<td>100 ms</td>
<td>860 ms</td>
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<tr>
<td>No Answer Warning</td>
<td>Admin</td>
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<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
<td>120000 ms</td>
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<td>Outgoing Address Register Type</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Outgoing Guard Time</td>
<td>Field</td>
<td>6000 ms</td>
<td>2000 ms</td>
<td>360 ms</td>
<td>1500 ms</td>
<td></td>
</tr>
<tr>
<td>Reconnect Call Feature Code</td>
<td>Admin</td>
<td></td>
<td></td>
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<td></td>
</tr>
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<td>Reconnect Call Request Signal</td>
<td>Admin</td>
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</tr>
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<td>Register Length</td>
<td>Admin</td>
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<td>5</td>
</tr>
<tr>
<td>Ringing Off Maximum</td>
<td>Admin</td>
<td></td>
<td>2400 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing Off Minimum</td>
<td>Admin</td>
<td></td>
<td>100 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Maximum</td>
<td>Admin</td>
<td></td>
<td>500 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Minimum</td>
<td>Admin</td>
<td></td>
<td>300 ms</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Seize Acknowledgment Timeout</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>T1 CAS Signaling Format</td>
<td>Admin</td>
<td></td>
<td></td>
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<td>Transfer Call Feature Code</td>
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<td></td>
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</tr>
<tr>
<td>Transfer Call Request Signal</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wink Start Delay</td>
<td>Field</td>
<td></td>
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</table>
### Table 23. Signaling types: template values (3) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>SL (Subscriber Loop) (E1)</th>
<th>U.K. CallStream (E1)</th>
<th>U.K. Exchange (E1)</th>
<th>U.K. Tie/DDI (E1)</th>
<th>EL7/CAS (Ericsson MD110) (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wink Start Duration</td>
<td>Field</td>
<td></td>
<td></td>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

### Table 24. Signaling types: template values (4)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>R2 Digital Line Signaling (E1)</th>
<th>TS003/P2 Line Signaling (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer Delay Time</td>
<td>Admin</td>
<td>260 ms</td>
<td>260 ms</td>
</tr>
<tr>
<td>Blocking Action</td>
<td>Field</td>
<td>Offhook</td>
<td>Offhook</td>
</tr>
<tr>
<td>Cadence Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
</tr>
<tr>
<td>Cadence Energy Minimum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>Cadence Off Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence Off Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>400 ms</td>
</tr>
<tr>
<td>Cadence On Time Maximum</td>
<td>Admin</td>
<td>600 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence On Time Minimum</td>
<td>Admin</td>
<td>400 ms</td>
<td>600 ms</td>
</tr>
<tr>
<td>Cadence Silence Maximum</td>
<td>Admin</td>
<td>-28 dBm</td>
<td>-28 dBm</td>
</tr>
<tr>
<td>CO Acknowledgment</td>
<td>Field</td>
<td>60 ms</td>
<td>60 ms</td>
</tr>
<tr>
<td>CO Off-Hook</td>
<td>Field</td>
<td>60 ms</td>
<td>60 ms</td>
</tr>
<tr>
<td>CO On-Hook</td>
<td>Field</td>
<td>200 ms</td>
<td>200 ms</td>
</tr>
<tr>
<td>Constant Energy Maximum</td>
<td>Admin</td>
<td>0 dBm</td>
<td>0 dBm</td>
</tr>
<tr>
<td>Constant Energy Minimum</td>
<td>Admin</td>
<td>-24 dBm</td>
<td>-24 dBm</td>
</tr>
<tr>
<td>Delay Start Delay</td>
<td>Field</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Delay Start Duration</td>
<td>Field</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Dial Pause</td>
<td>Admin</td>
<td>200 ms</td>
<td>200 ms</td>
</tr>
<tr>
<td>Dial Tone Qualify Time</td>
<td>Admin</td>
<td>200 ms</td>
<td>200 ms</td>
</tr>
<tr>
<td>Dial Tone (Detection) Timeout</td>
<td>Admin</td>
<td>8000 ms</td>
<td>8000 ms</td>
</tr>
</tbody>
</table>
Table 24. Signaling types: template values (4) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>R2 Digital Line Signaling (E1)</th>
<th>TS003/P2 Line Signaling (E1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Glare Detection Time</td>
<td>Field</td>
<td>40 ms</td>
<td>100 ms</td>
</tr>
<tr>
<td>Ground Flash</td>
<td>Field</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hangup Detection</td>
<td>Admin</td>
<td>Off</td>
<td>Off</td>
</tr>
<tr>
<td>Hook Flash</td>
<td>Admin</td>
<td>100 ms</td>
<td>100 ms</td>
</tr>
<tr>
<td>Incoming Address Register Type</td>
<td>Admin</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
</tr>
<tr>
<td>Incoming Guard Time</td>
<td>Field</td>
<td>100 ms</td>
<td>80 ms</td>
</tr>
<tr>
<td>No Answer Warning</td>
<td>Admin</td>
<td>120000 ms</td>
<td>120000 ms</td>
</tr>
<tr>
<td>Outgoing Address Register Type</td>
<td>Admin</td>
<td>Fixed Length</td>
<td>Fixed Length</td>
</tr>
<tr>
<td>Outgoing Guard Time</td>
<td>Field</td>
<td>1000 ms</td>
<td>120 ms</td>
</tr>
<tr>
<td>Reconnect Call Feature Code</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Reconnect Call Request Signal</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Register Length</td>
<td>Admin</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Ringing Off Maximum</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing Off Minimum</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Maximum</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ringing On Minimum</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Seize Acknowledgment Timeout</td>
<td>Field</td>
<td>200 ms</td>
<td>800 ms</td>
</tr>
<tr>
<td>T1 CAS Signaling Format</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transfer Call Feature Code</td>
<td>Admin</td>
<td>H</td>
<td>H</td>
</tr>
<tr>
<td>Transfer Call Request Signal</td>
<td>Admin</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wink Start Delay</td>
<td>Field</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wink Start Duration</td>
<td>Field</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Trunk interface templates

Templates are provided for the following trunk interfaces, and the supplied values for the parameters are shown in the following tables:

- E1
- E1 France
- E1 Italy
- E1 U.K.
- E1 ISDN
- E1 common channel signaling (CCS)
- E1 common channel signaling (CCS) U.K.
- E1 common channel signaling (CCS) Italy
- T1
- T1 (Aspect)
- T1 ISDN (XPACK)
- T1 common channel signaling (CCS)

Note: In the tables, a blank cell indicates that the parameter has no meaning for the protocol to which the template applies.

Table 25. Trunk interface: template values (1)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>E1</th>
<th>E1 France</th>
<th>E1 Italy</th>
<th>E1 U.K.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Answer Detect Time</td>
<td>Field</td>
<td>60 ms</td>
<td></td>
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<tr>
<td>Answer Detect Threshold</td>
<td>Field</td>
<td>-37 dBm</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>Automatically Re-Enable After Hardware Failure</td>
<td>Admin</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Backup Time and Erase after DTMF (1 unit = 20 ms)</td>
<td>Admin</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
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<tr>
<td>CCS Signaling Link Mode</td>
<td>Field</td>
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<td>None</td>
<td>None</td>
<td>None</td>
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<td>DP Receive Maximum Break</td>
<td>Field</td>
<td>100 ms</td>
<td>100 ms</td>
<td>60 ms</td>
<td>120 ms</td>
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<tr>
<td>DP Receive Minimum Break</td>
<td>Field</td>
<td>20 ms</td>
<td>20 ms</td>
<td>40 ms</td>
<td>40 ms</td>
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<td>System Parameter</td>
<td>Access Level</td>
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<td>E1 France</td>
<td>E1 Italy</td>
<td>E1 U.K.</td>
</tr>
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<td>----------</td>
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<tr>
<td>DP Receive Maximum Make</td>
<td>Field</td>
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<td>60 ms</td>
<td>60 ms</td>
<td>80 ms</td>
</tr>
<tr>
<td>DP Receive Minimum Make</td>
<td>Field</td>
<td>20 ms</td>
<td>20 ms</td>
<td>40 ms</td>
<td>20 ms</td>
</tr>
<tr>
<td>DP Transmit Break</td>
<td>Field</td>
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<td>50 ms</td>
<td>50 ms</td>
<td>68 ms</td>
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<tr>
<td>DP Transmit Speed (pulse/sec)</td>
<td>Field</td>
<td>10</td>
<td>10</td>
<td>6</td>
<td>10</td>
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<td>DTMF Maximum Receive Level</td>
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<tr>
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<td>-43 dBm</td>
<td>-41 dBm</td>
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<td>DTMF Transmit Level, Low Frequency</td>
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<td>-10 dBm</td>
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<td>DTMF Transmit Level Twist</td>
<td>Field</td>
<td>2 dBm</td>
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<td>2 dBm</td>
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<td>Field</td>
<td>50 ms</td>
<td>70 ms</td>
<td>80 ms</td>
<td>70 ms</td>
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<tr>
<td>DTMF Transmit Speed (digits/sec)</td>
<td>Field</td>
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<td>8</td>
<td>6</td>
<td>7</td>
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<tr>
<td>E1 Framing Mode</td>
<td>Field</td>
<td>Double</td>
<td>Double</td>
<td>Double</td>
<td>Double</td>
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<tr>
<td>E1 Hit Filter (1 unit = 2 ms)</td>
<td>Field</td>
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<td>E1 Timeslot 0 Word</td>
<td>Field</td>
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<td>1101 1111</td>
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<td>E1 Timeslot 16 Word</td>
<td>Field</td>
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<td>0000 1011</td>
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<td>E1 Italy</td>
<td>E1 U.K.</td>
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<td>Hand Shake Threshold</td>
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<td>6000 ms</td>
<td>6000 ms</td>
<td>6000 ms</td>
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<td>Idle Channel Code</td>
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<td>E1</td>
<td>UK/Italy</td>
<td>UK/Italy</td>
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<tr>
<td>Interdigit Pause Receive</td>
<td>Field</td>
<td>600 ms</td>
<td>300 ms</td>
<td>400 ms</td>
<td>180 ms</td>
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<tr>
<td>Interdigit Pause Transmit</td>
<td>Field</td>
<td>1000 ms</td>
<td>350 ms</td>
<td>1000 ms</td>
<td>1000 ms</td>
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<td>ISDN T1-NFAS Support</td>
<td>Field</td>
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<td>No</td>
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<td>No</td>
</tr>
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<td>ISDN Trunk Identifier</td>
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<tr>
<td>Maximum Playback Level</td>
<td>Field</td>
<td>-15 dBm</td>
<td>-15 dBm</td>
<td>-10 dBm</td>
<td>-15 dBm</td>
</tr>
<tr>
<td>Maximum Silence Duration</td>
<td>Field</td>
<td>12000 ms</td>
<td>12000 ms</td>
<td>10000 ms</td>
<td>120000 ms</td>
</tr>
<tr>
<td>Maximum Silence Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-99 units</td>
<td>-99 units</td>
<td>-95 units</td>
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</tr>
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<td>MFR1 Receive Level</td>
<td>Field</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
</tr>
<tr>
<td>Minimum Speech Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
</tr>
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<td>Music Absolute Silence Threshold</td>
<td>Field</td>
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<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
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<td>Music Channels Maximum</td>
<td>Admin</td>
<td>1</td>
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Table 25. Trunk interface: template values (1) (continued)

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<tr>
<th>System Parameter</th>
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<th>E1</th>
<th>E1 France</th>
<th>E1 Italy</th>
<th>E1 U.K.</th>
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<td>Operating Status</td>
<td>Admin Defined Defined Defined Defined</td>
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<td>Record DTMF Level</td>
<td>Admin -43 dBm -41 dBm -43 dBm -43 dBm</td>
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<td>Send RAI</td>
<td>Field No No No No</td>
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<td>Settle Time</td>
<td>Field 2000 ms 2000 ms 2000 ms 2000 ms</td>
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<td>Signaling Trunk Identifier</td>
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<td>Admin Default France Telecom Italy PSTN CAS Mercury (UK)</td>
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<td>T1 Bit Robbing</td>
<td>Field</td>
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<td>T1 Framing Mode</td>
<td>Field</td>
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<td>T1 Hit Filter (1 unit = 1.5 ms)</td>
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<td>T1 Remote Alarm Format</td>
<td>Field</td>
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<td>Trunk Signaling Mode</td>
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<td>System Parameter</td>
<td>Access Level</td>
<td>E1 ISDN</td>
<td>E1 CCS</td>
<td>E1 CCS U.K.</td>
<td>E1 CCS Italy</td>
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<td>Answer Detect Time</td>
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<td>60 ms</td>
<td>60 ms</td>
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<td>Answer Detect Threshold</td>
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<td>-37 dBm</td>
<td>-37 dBm</td>
<td>-37 dBm</td>
<td>-37 dBm</td>
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<td>Automatically Re-Enable After Hardware Failure</td>
<td>Admin Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
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<td>Backup Time and Erase after DTMF (1 unit = 20 ms)</td>
<td>Admin 5 units</td>
<td>5 units</td>
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<td>5 units</td>
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<td>CCS Signaling Link</td>
<td>Admin Yes</td>
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<td>Yes</td>
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<td>Yes</td>
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<td>Admin Primary</td>
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<td>CCS Signaling Link Timeslot</td>
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<td>DP Receive Maximum Break</td>
<td>Field 100 ms</td>
<td>100 ms</td>
<td>120 ms</td>
<td>60 ms</td>
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</tr>
<tr>
<td>DP Receive Minimum Break</td>
<td>Field 20 ms</td>
<td>20 ms</td>
<td>40 ms</td>
<td>40 ms</td>
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<tr>
<td>DP Receive Maximum Make</td>
<td>Field 60 ms</td>
<td>60 ms</td>
<td>80 ms</td>
<td>60 ms</td>
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<tr>
<td>DP Receive Minimum Make</td>
<td>Field 20 ms</td>
<td>20 ms</td>
<td>20 ms</td>
<td>40 ms</td>
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</tr>
<tr>
<td>DP Transmit Break (ms)</td>
<td>Field 60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
<td>50 ms</td>
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<tr>
<td>DP Transmit Speed (pulses/sec)</td>
<td>Field 10 ms</td>
<td>10 ms</td>
<td>10 ms</td>
<td>10 ms</td>
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<tr>
<td>DTMF Maximum Receive Level</td>
<td>Admin 0 dBm</td>
<td>0 dBm</td>
<td>-7 dBm</td>
<td>-3 dBm</td>
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<td>System Parameter</td>
<td>Access Level</td>
<td>E1 ISDN</td>
<td>E1 CCS</td>
<td>E1 CCS U.K.</td>
<td>E1 CCS Italy</td>
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<td>-------------</td>
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<td>DTMF Minimum Receive Level</td>
<td>Admin</td>
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<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-24 dBm</td>
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<td>Field</td>
<td>-8 dBm</td>
<td>-8 dBm</td>
<td>-10 dBm</td>
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<td>DTMF Transmit Level Twist</td>
<td>Field</td>
<td>2 dBm</td>
<td>2 dBm</td>
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<td>2 dBm</td>
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<td>DTMF Transmit On</td>
<td>Field</td>
<td>50 ms</td>
<td>50 ms</td>
<td>70 ms</td>
<td>80 ms</td>
</tr>
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<td>DTMF Transmit Speed (digits/sec)</td>
<td>Field</td>
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<td>E1 Framing Mode</td>
<td>Field</td>
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<td>Double</td>
<td>Double</td>
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<td>E1 Hit Filter (1 unit = 2 ms)</td>
<td>Field</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
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<td>E1 Timeslot 0 Word</td>
<td>Field</td>
<td>1101 1111</td>
<td>1101 1111</td>
<td>1101 1111</td>
<td>1101 1111</td>
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<td>E1 Timeslot 16 Word</td>
<td>Field</td>
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<td>0000 1011</td>
<td>0000 1011</td>
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</tr>
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<td>Hand Shake Threshold</td>
<td>Field</td>
<td>6000 ms</td>
<td>6000 ms</td>
<td>6000 ms</td>
<td>6000 ms</td>
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<tr>
<td>Idle Channel Code</td>
<td>Field</td>
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<td>E1</td>
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<td>UK/Italy</td>
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<td>Interdigit Pause Receive</td>
<td>Field</td>
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<tr>
<td>Interdigit Pause Transmit</td>
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<td>1000 ms</td>
<td>1000 ms</td>
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<td>ISDN T1-NFAS Support</td>
<td>Field</td>
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<td>No</td>
<td>No</td>
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<td>ISDN Trunk Identifier</td>
<td>Admin</td>
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<td>Maximum Playback Level</td>
<td>Field</td>
<td>-15 dBm</td>
<td>-15 dBm</td>
<td>-15 dBm</td>
<td>-10 dBm</td>
</tr>
<tr>
<td>Maximum Silence Duration</td>
<td>Field</td>
<td>12000 ms</td>
<td>12000 ms</td>
<td>12000 ms</td>
<td>10000 ms</td>
</tr>
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</table>
Table 26. Trunk interface: template values (2) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>E1 ISDN</th>
<th>E1 CCS</th>
<th>E1 CCS U.K.</th>
<th>E1 CCS Italy</th>
</tr>
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<tbody>
<tr>
<td>Maximum Silence Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-99 units</td>
<td>-99 units</td>
<td>-99 units</td>
<td>-95 units</td>
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<tr>
<td>MFR1 Receive Level</td>
<td>Field</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
</tr>
<tr>
<td>Minimum Speech Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
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<tr>
<td>Music Absolute Silence Threshold</td>
<td>Field</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
</tr>
<tr>
<td>Music Channels Maximum</td>
<td>Admin</td>
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<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Operating Status</td>
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<td>Defined</td>
<td>Defined</td>
<td>Defined</td>
</tr>
<tr>
<td>Record DTMF Level</td>
<td>Admin</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
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<td>Send RAI</td>
<td>Field</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Settle Time</td>
<td>Field</td>
<td>2000 ms</td>
<td>2000 ms</td>
<td>2000 ms</td>
<td>2000 ms</td>
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<td>Signaling Trunk Identifier</td>
<td>Admin</td>
<td>—</td>
<td>—</td>
<td>—</td>
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</tr>
<tr>
<td>Switch Encoding Law</td>
<td>Field</td>
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<td>Default</td>
<td>Default</td>
<td>Default</td>
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<td>Default</td>
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<td>Italy PSTN CAS</td>
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<td>T1 Bit Robbing</td>
<td>Field</td>
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<td></td>
</tr>
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<td>T1 Framing Mode</td>
<td>Field</td>
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<td></td>
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<td></td>
</tr>
<tr>
<td>T1 Hit Filter (1 unit = 1.5 ms)</td>
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<tr>
<td>T1 Line Code</td>
<td>Field</td>
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<td>T1 Remote Alarm Format</td>
<td>Field</td>
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<tr>
<td>Trunk Interlock - EDL</td>
<td>Admin</td>
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<td>Disabled</td>
<td>Disabled</td>
<td>Disabled</td>
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<td>Trunk Signaling Mode</td>
<td>Field</td>
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<td>CCS-SP</td>
<td>CCS-SP</td>
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Appendix B. System parameter templates 591
Table 27. Trunk interface: template values (3)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>T1</th>
<th>T1 (Aspect)</th>
<th>T1 ISDN (XPACK)</th>
<th>T1 CCS</th>
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<tr>
<td>Answer Detect Time</td>
<td>Field</td>
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<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
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<tr>
<td>Answer Detect Threshold</td>
<td>Field</td>
<td>-37 dBm</td>
<td>-37 dBm</td>
<td>-37 dBm</td>
<td>-37 dBm</td>
</tr>
<tr>
<td>Automatically Re-Enable After Hardware Failure</td>
<td>Admin</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Backup Time and Erase after DTMF (1 unit = 20 ms)</td>
<td>Admin</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
<td>5 units</td>
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<tr>
<td>CCS Signaling Link</td>
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<td>None</td>
<td>Primary</td>
<td>None</td>
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<td>CCS Signaling Link Mode</td>
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<td>None</td>
<td>Primary</td>
<td>None</td>
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<td>CCS Signaling Link Name</td>
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<td>None</td>
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<td>CCS Signaling Link Timeslot</td>
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<td>DP Receive Maximum Break</td>
<td>Field</td>
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<td>100 ms</td>
<td>100 ms</td>
<td>100 ms</td>
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<td>DP Receive Minimum Break</td>
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<td>20 ms</td>
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<td>DP Receive Maximum Make</td>
<td>Field</td>
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<td>60 ms</td>
<td>60 ms</td>
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<td>DP Receive Minimum Make</td>
<td>Field</td>
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<td>20 ms</td>
<td>20 ms</td>
<td>20 ms</td>
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<tr>
<td>DP Transmit Break (ms)</td>
<td>Field</td>
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<td>60 ms</td>
<td>60 ms</td>
<td>60 ms</td>
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<td>DP Transmit Speed (pulses/sec)</td>
<td>Field</td>
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<td>0 dBm</td>
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<td>T1 (Aspect)</td>
<td>T1 ISDN (XPACK)</td>
<td>T1 CCS</td>
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</tr>
<tr>
<td>DTMF Minimum Receive Level</td>
<td>Admin</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
</tr>
<tr>
<td>DTMF Transmit Level, Low Frequency</td>
<td>Field</td>
<td>-7 dBm</td>
<td>-7 dBm</td>
<td>-8 dBm</td>
<td>-7 dBm</td>
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<td>DTMF Transmit Level Twist</td>
<td>Field</td>
<td>0 dBm</td>
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<td>0 dBm</td>
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<td>DTMF Transmit On</td>
<td>Field</td>
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<td>50 ms</td>
<td>50 ms</td>
<td>50 ms</td>
</tr>
<tr>
<td>DTMF Transmit Speed (digits/sec)</td>
<td>Field</td>
<td>10</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>E1 Framing Mode</td>
<td>Field</td>
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<td></td>
<td></td>
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</tr>
<tr>
<td>E1 Hit Filter (1 unit = 2 ms)</td>
<td>Field</td>
<td></td>
<td></td>
<td></td>
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</tr>
<tr>
<td>E1 Timeslot 0 Word</td>
<td>Field</td>
<td></td>
<td></td>
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</tr>
<tr>
<td>E1 Timeslot 16 Word</td>
<td>Field</td>
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<tr>
<td>Hand Shake Threshold</td>
<td>Field</td>
<td>6000 ms</td>
<td>6000 ms</td>
<td>6000 ms</td>
<td>6000 ms</td>
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<tr>
<td>Idle Channel Code</td>
<td>Field</td>
<td>T1</td>
<td>T1</td>
<td>T1</td>
<td>T1</td>
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<td>Field</td>
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<td>280 ms</td>
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<tr>
<td>Interdigit Pause Transmit</td>
<td>Field</td>
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<td>1000 ms</td>
<td>1000 ms</td>
<td>1000 ms</td>
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<td>No</td>
<td>No</td>
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<td>0</td>
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<td>Maximum Playback Level</td>
<td>Field</td>
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<td>-15 dBm</td>
<td>-15 dBm</td>
<td>-15 dBm</td>
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<tr>
<td>Maximum Silence Duration</td>
<td>Field</td>
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<td>12000 ms</td>
<td>12000 ms</td>
<td>12000 ms</td>
</tr>
</tbody>
</table>
### Table 27. Trunk interface: template values (3) (continued)

<table>
<thead>
<tr>
<th>System Parameter</th>
<th>Access Level</th>
<th>T1</th>
<th>T1 (Aspect)</th>
<th>T1 ISDN (XPACK)</th>
<th>T1 CCS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Silence Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-99 units</td>
<td>-99 units</td>
<td>-99 units</td>
<td>-99 units</td>
</tr>
<tr>
<td>MFR1 Receive Level</td>
<td>Field</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
</tr>
<tr>
<td>Minimum Speech Level (1 unit = 0.5 dBm)</td>
<td>Field</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
<td>-81 units</td>
</tr>
<tr>
<td>Music Absolute Silence Threshold</td>
<td>Field</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
<td>-50 dBm</td>
</tr>
<tr>
<td>Music Channels Maximum</td>
<td>Admin</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Operating Status</td>
<td>Admin</td>
<td>Defined</td>
<td>Defined</td>
<td>Defined</td>
<td>Defined</td>
</tr>
<tr>
<td>Record DTMF Level</td>
<td>Admin</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
<td>-43 dBm</td>
</tr>
<tr>
<td>Send RAI</td>
<td>Field</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Settle Time</td>
<td>Field</td>
<td>15000 ms</td>
<td>15000 ms</td>
<td>2000 ms</td>
<td>15000 ms</td>
</tr>
<tr>
<td>Signaling Trunk Identifier</td>
<td>Admin</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Switch Encoding Law</td>
<td>Field</td>
<td>Default</td>
<td>Default</td>
<td>Default</td>
<td>Default</td>
</tr>
<tr>
<td>Switch Type</td>
<td>Admin</td>
<td>Default</td>
<td>Default</td>
<td>Default</td>
<td>Default</td>
</tr>
<tr>
<td>T1 Bit Robbing</td>
<td>Field</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>T1 Hit Filter (1 unit = 1.5 ms)</td>
<td>Field</td>
<td>7 units</td>
<td>7 units</td>
<td>7 units</td>
<td>7 units</td>
</tr>
<tr>
<td>T1 Line Code</td>
<td>Field</td>
<td>AMI</td>
<td>AMI</td>
<td>B8ZS</td>
<td>B8ZS</td>
</tr>
<tr>
<td>T1 Remote Alarm Format</td>
<td>Field</td>
<td>RAI via bit 2=0 in every channel</td>
<td>RAI via bit 2=0 in every channel</td>
<td>RAI via FS bit of frame 12</td>
<td>RAI via bit 2=0 in every channel</td>
</tr>
<tr>
<td>Trunk Interlock - EDL</td>
<td>Admin</td>
<td>Disabled</td>
<td>Disabled</td>
<td>Disabled</td>
<td>Disabled</td>
</tr>
<tr>
<td>Trunk Signaling Mode</td>
<td>Field</td>
<td>CAS</td>
<td>CAS</td>
<td>ISDN</td>
<td>CCS-SP</td>
</tr>
</tbody>
</table>
Call progress tone templates

If you use Pack Configuration to set up your telephony environment, the call progress tones that are in the required template are automatically selected for use in the country or region and for the switch that you specify. You can find out what values have been set by selecting the Direction & Call Progress Tones button in the Pack Configuration window (see “Displaying call progress tone values” on page 108).

If you need to change these values, the values in the Call Progress Tone parameters group must be edited, using System Configuration. This must be done by your IBM representative. For more information, see “Setting call progress tone parameters for outbound dialing” on page 101.

- “Call progress tones: Belgium” on page 596
- “Call progress tones: Brazil” on page 597
- “Call progress tones: Finland” on page 597
- “Call progress tones: France” on page 598
- “Call progress tones: Germany” on page 599
- “Call progress tones: Italy” on page 599
- “Call progress tones: the Netherlands” on page 600
- “Call progress tones: Spain” on page 600
- “Call progress tones: United Kingdom” on page 601
- “Call progress tones: U.S. and Canada” on page 602
  - “Special information tones: U.S. and Canada” on page 603
  - “PBX-specific tones” on page 606.

About the tables

One template exists for each tone. In these tables, the tone template names are in the leftmost column (Tone Name), and the system parameter names are across the column headings. The Tone Name is used only in these tables.

Each tone can have up to three frequency components, with accompanying levels and times on and off. Values for each of these are listed one above another. The column headed \( n \) indicates the frequency component (1, 2, or 3).

The columns headed Nom show the nominal values that are specified for the tone by the relevant authority or switch manufacturer. The nominal values are shown for information only. The templates contain only the extremes (Min and Max) of the range that defines the tone.

The column headed Tone Type corresponds to the value that is stored in the System: Progress tone type system variable (SV175). The values returned and the equivalent tone types are as follows:

SV175 Tone Type
The identifier that is stored in the System : Progress tone ID system variable (SV176) is not shown in these tables, but it is shown in the Call Progress Tones window (see “Displaying call progress tone values” on page 108).

**Call progress tones: Belgium**

*Table 28. Tone parameter values (Belgium)*

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level n (dBm)</th>
<th>Time On n (ms)</th>
<th>Time Off n (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1 410 450 491</td>
<td>-30 -9</td>
<td>Continuous tone</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1 800 900 1250</td>
<td>-30 -9</td>
<td>300 388 460</td>
<td>300 388 460</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 800 1020 1250</td>
<td>-30 -9</td>
<td>300 388 360</td>
<td>—</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 800 1140 1250</td>
<td>-30 -9</td>
<td>300 388 460</td>
<td>—</td>
</tr>
<tr>
<td>Ring Tone</td>
<td>Ring</td>
<td>Ring</td>
<td>1 410 450 491</td>
<td>-40 -9</td>
<td>820 1000 1160</td>
<td>2520 3000 3460</td>
</tr>
<tr>
<td>Busy</td>
<td>Busy</td>
<td>Busy</td>
<td>1 410 450 491</td>
<td>-40 -9</td>
<td>100 150 180</td>
<td>100 150 180</td>
</tr>
</tbody>
</table>

1. International dial tone is a Special Information Tone (SIT) and consists of a sequence of three frequencies with no off-time between them.
### Call progress tones: Brazil

Table 29. Tone parameter values (Brazil)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1 387 425 463</td>
<td>-25 -17</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Continuous tone

| Ring Tone | Ring       | Ring      | 1 387 425 463    | -40 -17     | 820 1000 1160 | 3380 4000 4620 |

| Busy Tone | Busy       | Busy      | 1 387 425 463    | -40 -17     | 400 500 580   | 400 500 580   |

### Call progress tones: Finland

Table 30. Tone parameter values (Finland)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1 387 425 463</td>
<td>-30 -9</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Continuous tone

| Ring Tone | Ring       | Ring      | 1 387 425 463    | -40 -9      | 820 1000 1160 | 3380 4000 4620 |

| Busy Tone | Busy       | Busy      | 1 387 425 463    | -40 -9      | 220 300 360   | 220 300 360   |
## Call progress tones: France

*Table 31. Tone parameter values (France)*

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1 250 440 485</td>
<td>-40 0</td>
<td>Continuous tone</td>
<td></td>
</tr>
<tr>
<td>Ring Tone</td>
<td>Ring</td>
<td>Ring</td>
<td>1 385 440 485</td>
<td>-40 0</td>
<td>1340 1500 1660</td>
<td>3340 3500 3660</td>
</tr>
<tr>
<td>Busy</td>
<td>Busy</td>
<td>Busy</td>
<td>1 385 440 485</td>
<td>-40 0</td>
<td>400 500 600</td>
<td>400 500 600</td>
</tr>
<tr>
<td>International Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1 275 330 385</td>
<td>-40 0</td>
<td>Continuous tone</td>
<td></td>
</tr>
<tr>
<td>Routing Tone</td>
<td>Routing</td>
<td>Other</td>
<td>1 406 440 474</td>
<td>-40 0</td>
<td>40 50 60</td>
<td>40 50 60</td>
</tr>
</tbody>
</table>
### Call progress tones: Germany

**Table 32. Tone parameter values (Germany)**

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
</tr>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1</td>
<td>372</td>
<td>435</td>
<td>503</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Min</td>
<td>Max</td>
<td>Min</td>
</tr>
<tr>
<td>Ring Tone</td>
<td>Ring</td>
<td>Ring</td>
<td>1</td>
<td>372</td>
<td>435</td>
<td>503</td>
</tr>
<tr>
<td>Busy</td>
<td>Busy</td>
<td>Busy</td>
<td>1</td>
<td>372</td>
<td>435</td>
<td>503</td>
</tr>
</tbody>
</table>

### Call progress tones: Italy

**Table 33. Tone parameter values (Italy)**

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
</tr>
<tr>
<td>Ring Tone</td>
<td>Ring</td>
<td>Ring</td>
<td>1</td>
<td>387</td>
<td>425</td>
<td>463</td>
</tr>
<tr>
<td>Network Busy</td>
<td>Network Busy</td>
<td>Busy</td>
<td>1</td>
<td>387</td>
<td>425</td>
<td>463</td>
</tr>
<tr>
<td>Busy</td>
<td>Busy</td>
<td>Busy</td>
<td>1</td>
<td>387</td>
<td>425</td>
<td>463</td>
</tr>
</tbody>
</table>
Call progress tones: the Netherlands

Table 34. Tone parameter values (Netherlands)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>n Min</td>
<td>Nom</td>
<td>Max</td>
<td>Min</td>
<td>Max</td>
<td>Min</td>
</tr>
<tr>
<td>Dial Tone 1</td>
<td>1</td>
<td>75</td>
<td>150</td>
<td>-23</td>
<td>8</td>
<td></td>
</tr>
<tr>
<td>Dial Tone 1</td>
<td>2</td>
<td>324</td>
<td>450</td>
<td>-23</td>
<td>8</td>
<td></td>
</tr>
<tr>
<td>Dial Tone 2</td>
<td>1</td>
<td>315</td>
<td>430</td>
<td>-23</td>
<td>8</td>
<td></td>
</tr>
<tr>
<td>Dial Tone 2</td>
<td>2</td>
<td>315</td>
<td>438</td>
<td>-40</td>
<td>-2</td>
<td>820</td>
</tr>
<tr>
<td>Busy</td>
<td>1</td>
<td>315</td>
<td>438</td>
<td>-40</td>
<td>-2</td>
<td>400</td>
</tr>
</tbody>
</table>

Call progress tones: Spain

Table 35. Tone parameter values (Spain)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>n Min</td>
<td>Nom</td>
<td>Max</td>
<td>Min</td>
<td>Max</td>
<td>Min</td>
</tr>
<tr>
<td>Dial Tone</td>
<td>1</td>
<td>320</td>
<td>400</td>
<td>-40</td>
<td>-9</td>
<td></td>
</tr>
<tr>
<td>Dial Tone</td>
<td>2</td>
<td>320</td>
<td>480</td>
<td>-40</td>
<td>-9</td>
<td></td>
</tr>
<tr>
<td>Tone Name</td>
<td>Tone Label</td>
<td>Tone Type</td>
<td>Frequency n (Hz)</td>
<td>Level (dBm)</td>
<td>Time On (ms)</td>
<td>Time Off (ms)</td>
</tr>
<tr>
<td>---------------------------</td>
<td>-----------------</td>
<td>-----------</td>
<td>------------------</td>
<td>-------------</td>
<td>--------------</td>
<td>---------------</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>n Min  Nom  Max</td>
<td>Min  Max</td>
<td>Min  Nom  Max</td>
<td>Min  Nom  Max</td>
</tr>
<tr>
<td>International Dial Tone</td>
<td>Dial, International</td>
<td>Dial</td>
<td>1  480  600  720</td>
<td>-40  -9</td>
<td>Continuous tone</td>
<td></td>
</tr>
<tr>
<td>Ring Tone</td>
<td>Ring</td>
<td>Ring</td>
<td>1  340  425  510</td>
<td>-40  -9</td>
<td>1240  1500  1740</td>
<td>2520  3000  3460</td>
</tr>
<tr>
<td>Busy Tone</td>
<td>Busy</td>
<td>Busy</td>
<td>1  340  435  510</td>
<td>-40  -9</td>
<td>140  200  240</td>
<td>140  200  240</td>
</tr>
</tbody>
</table>

Call progress tones: United Kingdom

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>n Min  Nom  Max</td>
<td>Min  Max</td>
<td>Min  Nom  Max</td>
<td>Min  Nom  Max</td>
</tr>
<tr>
<td>Dial Tone</td>
<td>Dial</td>
<td>Dial</td>
<td>1  319  350  382</td>
<td>-30  -9</td>
<td>Continuous tone</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2  400  440  480</td>
<td>-30  -9</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ring Part 1 Tone</td>
<td>Ring, Part 1</td>
<td>Ring</td>
<td>1  364  400  436</td>
<td>-40  -9</td>
<td>320  400  480</td>
<td>140  200  240</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2  410  450  491</td>
<td>-40  -9</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Table 36. Tone parameter values (United Kingdom) (continued)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
<td>Max</td>
<td>Min</td>
</tr>
<tr>
<td>Ring Part 2 Tone</td>
<td>Ring, Part 2</td>
<td>1</td>
<td>364</td>
<td>400</td>
<td>436</td>
<td>-40</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td>410</td>
<td>450</td>
<td>491</td>
<td>-40</td>
</tr>
<tr>
<td>Busy Tone</td>
<td>Busy</td>
<td>1</td>
<td>364</td>
<td>400</td>
<td>436</td>
<td>-40</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Busy Network Tone</td>
<td>Busy, Network</td>
<td>1</td>
<td>364</td>
<td>400</td>
<td>436</td>
<td>-40</td>
</tr>
</tbody>
</table>

Call progress tones: U.S. and Canada

Table 37. Tone parameter values (U.S./Canada)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
<td>Max</td>
<td>Min</td>
</tr>
<tr>
<td>Primary Dial Tone</td>
<td>Dial, Primary</td>
<td>1</td>
<td>319</td>
<td>350</td>
<td>382</td>
<td>-33</td>
</tr>
<tr>
<td>Dial</td>
<td></td>
<td>2</td>
<td>400</td>
<td>440</td>
<td>480</td>
<td>-33</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Continuous tone</td>
</tr>
</tbody>
</table>
Table 37. Tone parameter values (U.S./Canada) (continued)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secondary Dial Tone</td>
<td>Dial, Secondary</td>
<td>Dial</td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
<td>Max</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>478</td>
<td>525</td>
<td>572</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2</td>
<td>601</td>
<td>660</td>
<td>719</td>
</tr>
<tr>
<td>Ring Tone</td>
<td>RingRing</td>
<td>Ring</td>
<td>n</td>
<td>400</td>
<td>440</td>
<td>480</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2</td>
<td>437</td>
<td>480</td>
<td>523</td>
</tr>
<tr>
<td>Busy Line Tone</td>
<td>Busy, Line</td>
<td>Busy</td>
<td>n</td>
<td>437</td>
<td>480</td>
<td>523</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2</td>
<td>564</td>
<td>620</td>
<td>676</td>
</tr>
<tr>
<td>Busy Network Tone</td>
<td>Busy, Network</td>
<td>Network Busy</td>
<td>n</td>
<td>437</td>
<td>480</td>
<td>523</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2</td>
<td>564</td>
<td>620</td>
<td>676</td>
</tr>
</tbody>
</table>

**Special information tones: U.S. and Canada**

This section defines the terms used in Table 38 on page 604.

**Reorder - SIT1 (RO)**
This SIT indicates generalized network error conditions across intraLATA (local access and transport area) networks. An example returned announcement is: “We’re sorry, your call did not go through. Will you please try your call again?”.

**Vacant Code - SIT2 (VC)**
This SIT indicates that a number was dialled incorrectly. An example
returned announcement is “We're sorry, your call cannot be completed as dialled. Please check the number and dial again”.

No Circuit - SIT3 (NC')
This SIT indicates facility trouble that prevents calls from being completed. An example returned announcement is “We're sorry, due to telephone company facility trouble, your call cannot be completed at this time. Please try again later”.

Intercept - SIT4 (IC)
This SIT indicates that the number dialled has been disconnected. An example returned announcement is “We're sorry, you have reached a number that has been disconnected or is no longer in service. If you feel you have reached this recording in error, please check the number and try your call again”.

Reorder - SIT5 (RO')
This SIT indicates network error conditions relating to long-distance carriers. An example returned announcement is “We're sorry, due to network difficulties, your long-distance call cannot be completed at this time. Please try your call again later”.

No Circuit - SIT6 (NC")
This SIT indicates facility trouble that prevents calls from being completed. An example returned announcement is “We're sorry, all circuits are busy now. Will you please try again later”.

Ineffective Other - SIT7 (IO)
This SIT indicates that a call cannot be completed as dialled because of restrictions in the telephone being used. An example returned announcement is “We're sorry, you call cannot be completed as dialled from the phone you are using. Please read the instruction card and dial again”.

**Table 38. Special information tone (SIT) parameter values (U.S./Canada)**

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Tone Type</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>n Min Nom</td>
<td>Max</td>
<td>Min Max</td>
<td>Min Nom Max</td>
<td>Min Nom Max</td>
<td></td>
</tr>
<tr>
<td>Reorder</td>
<td>SIT1(RO')</td>
<td>Network Busy</td>
<td>1 873 914 955 -40 -9 220 274 320</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td></td>
<td>2 1365</td>
<td>1429 1493</td>
<td>-40 -9 340 380 460</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td></td>
<td>3 1620</td>
<td>1770 1857</td>
<td>-40 -9 340 380 460</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>Tone Name</td>
<td>Tone Label</td>
<td>Tone Type</td>
<td>Frequency n (Hz)</td>
<td>Level (dBm)</td>
<td>Time On (ms)</td>
<td>Time Off (ms)</td>
</tr>
<tr>
<td>----------------</td>
<td>------------</td>
<td>-----------</td>
<td>------------------</td>
<td>-------------</td>
<td>--------------</td>
<td>--------------</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>n Min Nom Max</td>
<td>Min Max</td>
<td>Min Nom Max</td>
<td></td>
</tr>
<tr>
<td>Vacant Code</td>
<td>SIT2 (VC)</td>
<td>Other</td>
<td>1 941 985 1029</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1309 1371 1433</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIT3 (NC')</td>
<td>Network Busy</td>
<td>1 941 985 1029</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1365 1429 1493</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIT4 (IC)</td>
<td>Other</td>
<td>1 873 914 955</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1309 1371 1433</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIT5 (RO')</td>
<td>Network Busy</td>
<td>1 941 985 1029</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1309 1371 1433</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIT6 (NC&quot;)</td>
<td>Network Busy</td>
<td>1 873 914 955</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1309 1371 1433</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIT7 (IO)</td>
<td>Other</td>
<td>1 873 914 955</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2 1365 1429 1493</td>
<td>-40 -9</td>
<td>220 274 320</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3 1620 1770 1857</td>
<td>-40 -9</td>
<td>340 380 460</td>
<td></td>
</tr>
</tbody>
</table>
Table 38. Special information tone (SIT) parameter values (U.S./Canada) (continued)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Frequency n (Hz)</th>
<th>Tone Type</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>n Min Nom Max</td>
<td></td>
<td>Min Max</td>
<td>Min Nom Max</td>
<td>Min Nom Max</td>
</tr>
</tbody>
</table>

1. Special Information Tones each consist of a sequence of three frequencies with no off-time between them.

PBX-specific tones

Table 39. Tones specific to AT&T and ROLM PBXs

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>n Min Nom Max</td>
<td></td>
<td>Min Max</td>
<td>Min Nom Max</td>
</tr>
</tbody>
</table>

- AT&T PBX Dial
  - Dial
    - 1 400 440 480 -27 -13 Continuous tone

- Rolm Dial Tone
  - Dial Tone Rolm 9015
    - 1 440 480 520 -33 -9 Continuous tone

- Rolm Dial Tone
  - Holding Dial Tone Rolm 9015
    - 1 260 300 340 -33 -9 Continuous tone
Table 39. Tones specific to AT&T and ROLM PBXs (continued)

<table>
<thead>
<tr>
<th>Tone Name</th>
<th>Tone Label</th>
<th>Frequency n (Hz)</th>
<th>Level (dBm)</th>
<th>Time On (ms)</th>
<th>Time Off (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rolm Ring</td>
<td>Ring, Rolm Ring</td>
<td>1</td>
<td>400 440 480</td>
<td>-40 -9</td>
<td>820 1000 1160</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td>437 480 523</td>
<td>-40 -9</td>
<td></td>
</tr>
<tr>
<td>Rolm Busy, Do Not Disturb</td>
<td>Busy, Rolm DND</td>
<td>1</td>
<td>701 770 839</td>
<td>-33 -9</td>
<td>400 500 580</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td>801 880 959</td>
<td>-33 -9</td>
<td></td>
</tr>
<tr>
<td>Rolm Error Tone</td>
<td>Error Tone Rolm 9005</td>
<td>1</td>
<td>280 300 320</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td>460 480 500</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td></td>
<td></td>
<td>3</td>
<td>280 300 320</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td>Rolm Error Tone</td>
<td>Error Tone Rolm 9005</td>
<td>1</td>
<td>460 480 500</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2</td>
<td>280 300 320</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td></td>
<td></td>
<td>3</td>
<td>460 480 500</td>
<td>-35 -5</td>
<td>180 250 320</td>
</tr>
<tr>
<td>Tone Name</td>
<td>Tone Label</td>
<td>Tone Type</td>
<td>Frequency n (Hz)</td>
<td>Level (dBm)</td>
<td>Time On (ms)</td>
</tr>
<tr>
<td>-------------------------</td>
<td>------------------</td>
<td>-----------</td>
<td>------------------</td>
<td>-------------</td>
<td>--------------</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>Min</td>
<td>Nom</td>
</tr>
<tr>
<td>Rolm Error Tone 2</td>
<td>Other</td>
<td></td>
<td>1</td>
<td>400</td>
<td>440</td>
</tr>
<tr>
<td></td>
<td>Rolm 9006</td>
<td></td>
<td>2</td>
<td>564</td>
<td>620</td>
</tr>
<tr>
<td></td>
<td>Other</td>
<td></td>
<td>3</td>
<td>400</td>
<td>440</td>
</tr>
<tr>
<td>Rolm Error Tone 2</td>
<td>Other</td>
<td></td>
<td>1</td>
<td>564</td>
<td>620</td>
</tr>
<tr>
<td></td>
<td>Rolm 9006</td>
<td></td>
<td>2</td>
<td>400</td>
<td>440</td>
</tr>
<tr>
<td></td>
<td>Other</td>
<td></td>
<td>3</td>
<td>564</td>
<td>620</td>
</tr>
</tbody>
</table>

1. Error tones alternate between two tones (300 Hz and 480 Hz for the Rolm 9005 or 440 Hz and 620 Hz for the Rolm 9006) at a rate of 250 ms, continuously, that is tone1-tone2-tone1-tone2... Two tritone entries are used for each type of tone to recognize tones starting with either the first or the second of the two tones.
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Glossary

The following terms and abbreviations are defined as they are used in the context of WebSphere Voice Response. If you do not find the term or abbreviation you are looking for, see IBM Dictionary of Computing, McGraw-Hill, 1994 or the AIX: Topic Index and Glossary, SC23–2513.

Special Characters

µ-law The companding algorithm that is used primarily in North America and Japan when converting from analog to digital speech data. (Compand is a contraction of compress and expand.) Contrast with A-law.

Numerics

2 B-channel transfer feature See Integrated Services Digital Network (ISDN) two B-channel transfer.

3270 host application An application on the IBM System/370™ System/390®, or AS/400® that interacts with terminals that support the 3270 data stream.

3270 script language See script language.

3270 server A function of WebSphere Voice Response that provides a software interface between WebSphere Voice Response and IBM System/370, System/390, or AS/400 architecture business applications that interact with terminals that support the 3270 data stream. Contrast with custom server.

5ESS (1) A Lucent Technologies switch. (2) The ISDN protocol that is used on the 5ESS switch. It provides 23 B-channels and a D-channel over a T1 trunk.

6312 Digital Trunk Telephony Adapter (DTTA) See Digital Trunk Telephony Adapter.

6313 Digital Trunk Telephony Adapter (DTTA) with Blind Swap Cassette (BSC) See Digital Trunk Telephony Adapter with Blind Swap Cassette.

A

A-law The companding algorithm that is used in Europe, Latin America, and other countries when converting from analog to digital speech data. (Compand is a contraction of compress and expand.) Contrast with µ-law.

access protocol A protocol that is used between an external subscriber and a switch in a telephone network.

ACD See automatic call distributor.

ACL See application connectivity link.

action See state table action.

Action Palette An area that contains folders and icons that can be selected to create state table actions.
Address Resolution Protocol (ARP)
In HACMP, the Internet communication protocol that dynamically maps Internet addresses to physical (hardware) addresses on local area networks. Limited to networks that support hardware broadcast.

The `usr/sbin/cluster/etc/clinfo.rc` script, which is invoked by the clinfo daemon whenever a network or node event occurs, updates the system ARP cache. This ensures that the IP addresses of all cluster nodes are updated after an IP address takeover. The script can be further customized to handle site-specific needs.

administrator profile
Data that describes a WebSphere Voice Response user. Information that is in an administrator profile includes ID, password, language preference, and access privileges.

ADSI
See `analog display services interface`.

ADSI telephone
A “smart” telephone that can interpret and return ADSI data.

advanced intelligent network (AIN)
A telephone network that expands the idea of the `intelligent network (IN)` to provide special services more efficiently; for example, by giving users the ability to program many of the services themselves.

AIN
See `advanced intelligent network`.

alarm
Any condition that WebSphere Voice Response thinks worthy of documenting with an `error message`. Strictly, the term `alarm` should include only red (immediate attention) and yellow (problem condition), but it is also used to refer to green (a red or yellow message has been cleared) and white (information) conditions. Contrast with `alert`.

alert
A message that is sent to a central monitoring station, as the result of an alarm. Contrast with `alarm`.

alternate mark inversion (AMI)
A T1 line coding scheme in which binary 1 bits are represented by alternate positive and negative pulses and binary 0 bits by spaces (no pulse). The purpose is to make the average dc level on the line equal to zero.

AMI
See `alternate mark inversion`.

analog
Data in the form of continuously variable signals, such as voice or light signals.

analog display services interface (ADSI)
A Bellcore signaling protocol that is used with existing voice networks. ADSI supports analog transmission of voice and text-based information between a host or switch, voice mail system, service bureau, or similar, and a subscriber's ADSI-compatible screen telephone. A single voice-grade telephony channel is shared between voice and data, using a technique by which the channel is taken over for the transmission of modem-encoded data.

ANI
See `automatic number identification`.

annotation
In speech recognition, an alphanumeric string that is used to mark a grammar when it is defined. When the grammar is used in an
application, both the word and the alphanumeric string are returned to the application.

**announcement-only greeting**
In voice mail, a *greeting* that does not give the *caller* a chance to leave a *voice message*.

**application**
A (usually) customer-written program or set of programs that might consist of one or more state tables or custom servers that are running on WebSphere Voice Response, with associated voice segments. See [voice application].

**application connectivity link (ACL)**
A service that transmits out-of-band information between WebSphere Voice Response and the Siemens Hicom 300 switch.

**application profile**
Data that describes initial actions that are to be performed when the telephone is answered. Information in an application profile indicates to the channel process which state table to load.

**application server interface (ASI)**
The principal software component of WebSphere Voice Response that manages the real-time channel processing.

**application server platform (ASP)**
A platform that is used for Web and voice applications for e-business. See [application server interface], [application server platform].

**audio name**
The audible name that relates to a specific application profile ID and mailbox.

**auto-attendant**
Automated attendant. A voice application that answers incoming calls and asks callers which number or other service they would like.

**automatic call distributor (ACD)**
A telephone system feature that automatically queues and processes inbound calls according to predefined rules. For example, a call might be routed to the agent whose line has been idle longest.

**automatic number identification (ANI)**
A service available in the U.S. that provides the telephone number of the calling party. It is generated by the caller’s originating central office switch, sent to a telephone network carrier if required, then sent directly either to a switch or to a voice processing system.

**autostubbing**
A state table icon view utility that automatically converts lines into stubs when they cross a specified number of columns.

**B**

**B8ZS** Bipolar with 8-zero substitution. A T1 line code that is required for 64Kb channels such as ISDN.

**B-channel**
See [bearer channel]. See also [Integrated Services Digital Network (ISDN)].

**background music**
Any audio data that is to be played on a *music channel*.

**barge-in**
The capability that allows a prompt to be interrupted by an utterance.
that is then passed to a speech recognizer. See also cut-through channel.

**baseforms**
The set of phonetic pronunciations that are associated with a grammar. In WebSphere Voice Server, the IBM dictionary of pronunciations is used.

**basic rate interface (BRI)**
The means of ISDN access that is normally used by private subscribers. It provides two B-channels of 64 Kb per second and one D-channel of 16 Kb per second for signaling. This is often known as 2B+D. Contrast with primary rate interface (PRI).

**beans**
Java beans with which you can build voice applications to use the services of WebSphere Voice Response on any platform.

**bearer channel**
In an ISDN interface, a duplex channel for transmitting data or digital voice between the terminal and the network. The B-channel operates at 64 Kb per second.

**bearer service**
The type of service that defines how an ISDN connection will be used. Typical bearer services are speech telephony, 64 Kb per second data, and high-quality speech.

**blind transfer**
A type of call transfer in which the call is routed to another extension and the original call is ended. No check is made to determine whether the transferred call is answered or if the number is busy. Contrast with screened transfer.

**bnf**
Abbreviation for Backus-Naur Form, which is used to describe the syntax of a given language and its notation. In speech recognition, a special adaptation of grammar representation that is specified by Speech Recognition Control Language (SRCL) (pronounced “circle”).

**bos**
Base Operating System.

**bps**
bits per second.

**BRI**
See basic rate interface.

**bridge**
See DVT bridge.

**British Approvals Board for Telecommunications**
The British standards organization that is responsible for approval of equipment that is to be attached to the PSTN.

**cadence**
The modulated and rhythmic recurrence of an audio signal. For example, a series of beeps or a series of rings.

**call**
Telephone call. Often used to mean a single run-time instance of a voice application.

**call center**
A central point at which all inbound calls are handled by a group of individuals in a controlled sequential way. Call centers are usually a front end to a business such as airline ticketing or mail order.

**Call Control eXtensible Markup Language (CCXML)**
Language designed to provide telephony call control support for VoiceXML or other dialog systems.
Refer to the CCXML forum web site at http://www.w3.org/TR/ccxml

call forwarding
The process of sending incoming calls to a different number.

called party
Any person, device, or system that receives a telephone call. Contrast with caller.

caller
(1) Any person, device, or system that makes a telephone call. (2) Often used to refer to any user of a voice application, although WebSphere Voice Response might have made an outbound call and the user is really the called party. (3) In voice mail, any person who makes a telephone call to a subscriber. Contrast with user.

calling line identification presentation (CLIP)
An ISDN supplementary service that advises the called party of the caller’s number, for example, by displaying it on a telephone display panel.

CallPath
Software that provides basic computer-telephony integration (CTI) enablement and comprehensive CTI functionality. This includes access to, and management of, inbound and outbound telecommunications.

call session
The sequence of events that occurs from the time a call is started to the time all activities related to answering and processing the call are completed.

call transfer
A series of actions that directs a call to another telephone number. See also dual-line call transfer.

CAS
See channel associated signaling.

cascading resources
Resources that can be taken over by more than one node. A takeover priority is assigned to each configured cluster resource group in a per-node way. In the event of a takeover, the node with the highest priority gets the resource group. If that node is unavailable, the node with the next-highest priority gets the resource group, and so on.

CAS tone
Customer Premise Equipment Alerting Signal tone. In ADSI, this tone is sent to the ADSI telephone to switch the phone to data mode.

CBX
See computerized branch exchange.

CCH
See Comité de Coordination de l’Harmonisation.

CCITT
See Comité Consultatif International Télégraphique et Téléphonique.

CCS
See common channel signaling (CCS).

central office (CO)
A telephone switching system that resides in the telephone service provider’s network. Different types of central office switches exist, depending upon the role of the switch in the telephone network. Commonly, a central office switch connects customer lines to other customer lines or trunks, and is the point at which local subscriber lines end for switching to other lines or trunks.

central registry
A component of the Licence Use
Management network topology. A server's database that logs requests for licenses, upgrades for licenses, and journals all license activity in a tamper-proof auditable file.

**CEPT**  
See [Conference Européenne des Administrations des Postes et Télécommunications](https://wwwCEPT.org).

**CGI**  
See [Common Gateway Interface](https://www.ISO.org).

**channel**  
One of the 24 channels that are on a T1 trunk, or one of the 30 channels that are on an E1 trunk. See also [speech recognition session](https://www.ACM.org), [music](https://www.Music.org).

**channel-associated signaling (CAS)**  
A method of communicating telephony supervisory or line signaling (on-hook and off-hook) and address signaling on T1 and E1 digital links. The signaling information for each traffic (voice) channel is transmitted in a signaling channel that is permanently associated with the traffic channel. On T1 links, supervisory signaling is sent in the traffic channel by using [robbed-bit signaling](https://www.ISO.org) (RBS). On E1 links, a separate channel is used to send signaling. Address signaling can be transmitted either in the signaling channel (out-of-band) or in the traffic channel (in-band).  
Contrast with [common channel signaling (CCS)](https://www.ISO.org).

**channel bank**  
A device that converts an analog line signal to a digital trunk signal.

**channel number**  
The identifying number that is assigned to a licensed channel on the T1 or E1 trunk that connects WebSphere Voice Response to the switch, channel bank, or channel service unit.

**channel process (CHP)**  
The AIX process that runs the logic of the state table; each active caller session has one active channel process.

**channel service unit (CSU)**  
A device that is used to connect a digital phone line to a multiplexer, a channel bank, or directly to another device that generates a digital signal. A CSU performs specific line-conditioning and equalization functions, and responds to loopback commands that are sent from the CO.

**CHP**  
See [channel process](https://www.ISO.org).

**CIC**  
See [circuit identification code](https://www.ISO.org).

**CICS**  
See [customer information control system](https://www.ISO.org).

**circuit identification code (CIC)**  
A 12-bit number that identifies a trunk and channel on which a call is carried.

**clear message**  
A message that is displayed by WebSphere Voice Response to tell the operator that a red or yellow error message has been cleared.

**client node**  
In a single system image (SSI), a WebSphere Voice Response system that handles interactions with callers. A client node must have a telephony connection. It does not store application or voice data; it gets data from the server node of the SSI.
CLIP  See calling line identification presentation.

cluster
Loosely-coupled collection of independent systems (nodes) that are organized into a network to share resources and to communicate with each other. HACMP defines relationships among cooperating systems where peer cluster nodes provide the services that a cluster node offers if that node cannot do so.

cluster configuration
User definition of all cluster components. Component information is stored in the Object Data Manager. Components include cluster name and ID, and information about member nodes, adapters, and network modules.

CO  See central office.

codec  Refers to adapters that compress and decompress video files. The letters "codec" represent "compression/decompression"; in the past, they represented "coder/decoder."

Comité de Coordination de l'Harmonization
The CEPT committee responsible for standards.

Comitato Elettrotecnico Italiano
The Italian standards organization responsible for signaling protocols.

Comité Consultatif International Télégraphique et Téléphonique (CCITT)
This organization has been renamed and is now known as the International Telecommunications Union - Telecommunication Standardization Sector (ITU-T).

common channel signaling (CCS)
A method of communicating telephony information and line signaling events (for example, call setup and call clearing) on a dedicated signaling channel. The signaling channel is either a predefined channel on an E1 or T1 digital link, or a completely separate link between the switch and WebSphere Voice Response. For data integrity and reliability, the information is usually communicated using a data link protocol. The telephone information and line signaling events are sent as data packets. SS7 and ISDN are common-channel signaling protocols. Contrast with channel associated signaling.

Common Gateway Interface (CGI)
An interface to programs that provide services on the world wide Web.

compiled grammar file
A grammar in binary format that was built by the WebSphere Voice Server grammar development tools.

compound license
In License Use Management, a type of license that allows a system administrator to generate license passwords for a given number of licenses. A compound license can generate either nodelocked or non-nodelocked licenses, but not both.

computer-telephony integration (CTI)
The use of a general-purpose computer to issue commands to a telephone switch to transfer calls and provide other services. Typically, CTI is used in call centers.
computerized branch exchange (CBX)
A computer-driven, digital communications controller that provides telephone communication between internal stations and external networks.

Conférence Européenne des Administrations des Postes et Télécommunications (CEPT)
European Conference of Postal and Telecommunications Administrations.

configuration parameter
A [variable] that controls the behavior of the system or the behavior of all applications that are running on the system. See parameter file, system parameter.

context parameter
A window that lists the names of all existing objects of the same type.

context
A set of one or more grammars that is enabled and used during a recognition action. The grammars are specified by a FILELIST file. Parameters that influence the recognition, such as the maximum initial silence period and the ending silence period, are also defined by the context. More than one context can be enabled for a recognition.

context name
The name given to a context in a context profile that is used for WebSphere Voice Server.

context profile
Describes to the WebSphere Voice Server process which contexts should be loaded into an engine. A WebSphere Voice Response for Windows application specifies which context profiles to load into the engine it has reserved.

context type
Indicates to the recognition engine how to interpret the grammar file. Possible types are: VOCAB_FILE, GRAMMAR_FILE, TEXT, MNR_FILE, MNR, PERSONAL_FILE, PERSONAL_WDS, BASEFORM_FILE.

continuous speech recognition
Recognition of words that are spoken in a continuous stream. Unlike isolated or discrete word recognition, users do not have to pause between words.

conversation
See speech recognition session.

CPE See customer premises equipment.

CSU See channel service unit.

CTI See computer-telephony integration.

customer information control system (CICS)
A licensed program that enables transactions that are entered at remote workstations to be processed concurrently by user-written application programs. It includes facilities for building, using, and maintaining databases.

custom server
A C language or C++ language program that provides data manipulation and local or remote data stream, database, or other services that are additional to those that the state table interface provides. Custom servers provide an interface between WebSphere
Voice Response and business applications, functions, or other processes to give callers access to business information and voice processing functions such as speech recognition.

customer premises equipment (CPE)
Telephony equipment that is on the premises of a business or domestic customer of the telephone company. An example is a private branch exchange (PBX).

cut-through channel
A channel of voice data that has been passed through echo-cancellation algorithms. The channel provides echo-cancelled voice data that can then be used by the engine in a recognition attempt. This is similar to **barge-in**.

D

daemon
In the AIX operating system, a program that runs unattended to perform a standard service.

database server node
In a single system image (SSI), a WebSphere Voice Response system that contains the WebSphere Voice Response DB2 database. This is usually the same node as the voice server node.

DBIM The internal database manager of WebSphere Voice Response.

DBS The database server of WebSphere Voice Response.

DCBU See **D-channel backup**

D-channel
See **delta channel**

D-channel backup (DCBU)
An ISDN NFAS configuration where two of the T1 facilities have a D-channel, one of which is used for signaling, and the other as a backup if the other fails. See also **non-facility associated signaling**.

DDI See **direct inward dialing**

DDS See **production system**

delay start
A procedure that is used with some channel-associated signaling protocols to indicate when a switch or PABX is ready to accept address signaling. After seizure, the switch sends off-hook until it is ready to accept address signaling, at which time it sends on-hook. Contrast with **immediate start** and **wink start**.

delta channel
In an ISDN interface, the D-channel or delta channel carries the signaling between the terminal and the network. In a basic rate interface, the D-channel operates at 16 Kb per second. In a primary rate interface, the D-channel operates at 64 Kb per second.

destination point code (DPC)
A code that identifies the signaling point to which an MTP signal unit is to be sent. Unique in a particular network.

development system
A WebSphere Voice Response system that is not used to respond to, or make, “live” calls; it is used only to develop and test applications. Contrast with **production system**.

dial
To start a telephone call. In telecommunication, this action is
performed to make a connection between a terminal and a telecommunication device over a switched line.

dial by name
To press the keys that are related to subscribers’ names instead of to their telephone numbers or extensions.

dialed number identification service (DNIS)
A number that is supplied by the public telephone network to identify a logical called party. For example, two toll-free numbers might both be translated to a single real number. The DNIS information distinguishes which of the two toll-free numbers was dialed.

dialog box
A secondary window that presents information or requests data for a selected action.

dial tone
An audible signal (call progress tone) that indicates that a device such as a PABX or central office switch is ready to accept address information (DTMF or dial pulses).

DID
See [direct inward dialing](#).

digital signal processing (DSP)
A set of algorithms and procedures that processes electronic signals after their conversion to digital format. Because of the specific mathematical models that are required to perform this processing, specialized processors are generally used.

Digital Subscriber signaling System Number 1 (DSS1)
A signaling protocol that is used between ISDN subscriber equipment and the network. It is carried on the ISDN D-channel. ITU-T recommendations Q.920 to Q.940 describe this protocol.

Digital Trunk Ethernet Adapter (DTEA)
A Radysis adapter card that provides the audio streaming (RTP) interface between the WebSphere Voice Response internal H.100 bus and Ethernet for a maximum of 120 channels using uncompressed (G.711) voice, and compressed G.723.2 and G.729A compressed voice.

Digital Trunk No Adapter (DTNA)
A device driver that supports uncompressed (G.711) voice RTP streaming.

Digital Trunk Telephony Adapter (DTTA)
The IBM Quad Digital Trunk Telephony PCI Adapter. In WebSphere Voice Response, this adapter is known as a DTTA. It allows you to connect directly to the telephony network from a pSeries computer without the need for an external pack.

Digital Trunk Telephony Adapter (DTTA) with Blind Swap Cassette (BSC)
The IBM Quad Digital Trunk Telephony PCI Adapter. In WebSphere Voice Response, this adapter is known as a DTTA. It allows you to connect directly to the telephony network from a pSeries computer without the need for an external pack. This DTTA includes a short Blind Swap Cassette (BSC) which is required for installing the DTTA in machines that use the BSC (for example, the pSeries 650–6M2).
diphone
A transitional phase from one sound to the next that is used as a building block for speech synthesis. Typically, between one thousand and two thousand diphones exist in any national language.

direct dial in (DDI)
See direct inward dialing.

direct inward dialing (DID)
A service that allows outside parties to call directly to an extension of a PABX. Known in Europe as direct dial in (DDI).

direct speech recognition
Identification of words from spoken input that are read directly from the telephony channel. Contrast with indirect speech recognition.

DirectTalk bean
One of the beans that is provided with WebSphere Voice Response. It provides access from a voice application to simple call control functions: waiting for a call, making an outgoing call, handing a call over to another application, and returning a call when finished.

discrete word recognition
Identification of spoken words that are separated by periods of silence, or input one at a time. Contrast with continuous speech recognition.

disconnect
To hang up or terminate a call.

Distributed Voice Technologies (DVT)
A component of WebSphere Voice Response that provides an interface to allow you to integrate your own voice technology (such as a speech recognizer) with your WebSphere Voice Response system.

distribution list
In voice mail, a list of subscribers to whom the same message can be sent.

DMS100
(1) A Northern Telecom switch. (2) The custom ISDN protocol that is run on the DMS100 switch, providing 23 B-channels and a D-channel over a T1 trunk.

DNIS
See dialed number identification service.

double-trunking
See trombone.

down
The condition in which a device is unusable as a result of an internal fault or of an external condition, such as loss of power.

downstream physical unit (DSPU)
Any remote physical unit (data link, storage, or input/output device) that is attached to a single network host system.

DPC
See destination point code.

drop-in grammar
A set of precompiled grammar rules that can be used by an application-specific grammar to improve the recognition performance.

DSP
See digital signal processing.

DSPU
See downstream physical unit.

DSS1
See Digital Subscriber signaling System Number 1.

DTMF
See dual-tone multifrequency.

DTEA
See Digital Trunk Ethernet Adapter.

DTNA
See Digital Trunk No Adapter.
**DTTA** See *Digital Trunk Telephony Adapter*.

**dtuser** The name of the AIX account that is set up during the installation process for the use of all users of WebSphere Voice Response.

**dual-line call transfer**
A call transfer method in which the primary and secondary lines remain bridged until a call is completed. (Also known as tromboning: see *trombone*).

**dual-tone multifrequency (DTMF)**
The signals are sent when one of the telephone keys is pressed. Each signal is composed of two different tones.

**DVT** See *Distributed Voice Technologies*.

**DVT bridge**
The interface between a voice technology component (such as a speech recognizer) and the DVT server. A bridge must exist for each technology that you want to integrate with DVT.

**DVT_Client2**
A WebSphere Voice Response custom server that passes commands and data to DVT_Server.

**DVT interface**
A WebSphere Voice Response programming interface that is used by a DVT bridge. It enables integration of voice applications with *Distributed Voice Technologies* to provide functions such as speech recognition.

**DVT_Server**
A component of DVT that allocates and manages system resources in response to requests from DVT_Client2.

**DVT service**
The combination of a voice application, a DVT bridge, and a voice technology that allows a caller to interact with your business.

**dynamic vocabulary**
A vocabulary that is defined while an application is running.

**E**

**E&M** A channel-associated signaling protocol in which signaling is done using two leads: an M-lead that transmits battery or ground and an E-lead that receives open or ground.

**E1** A digital trunking facility standard that is used in Europe and elsewhere. It can transmit and receive 30 digitized voice or data channels. Two additional channels are used for synchronization, framing, and signaling. The transmission rate is 2048 Kbps per second. Contrast with T1.

**echo cancelation**
A filter algorithm that compares a copy of the voice data that is being sent to a caller, with the voice data that is received from the caller. Any echo of the sent data is removed before the received data is sent on, for example, to a speech recognizer.

**edge** See *result*.

**EDL** See *exchange data link*.

**emulation**
The imitation of all or part of one computer system by another, so that the imitating system accepts the same data, runs the same programs, and gets the same results as the imitated computer system does.
**endpoint**  
In Voice over Internet Protocol, a place where calls are originated and ended.

**engine**  
A speech recognition process that accepts voice data as input and returns the text of what was said as output. It is the process that performs the recognition.

**engine type**  
Each engine must be configured with a specific type. The type is a textual tag that is associated with a specific engine and does not change the operation or functionality of the engine.

**error message**  
Any message that is displayed by WebSphere Voice Response in the System Monitor as an alarm and optionally written to the WebSphere Voice Response error log, or to the AIX error log (as an alert). Strictly, the term error message should include only red (immediate attention) and yellow (problem situation) messages, but it is also used to refer to green (a red or yellow message has been cleared) and white (informational) messages.

**Ethernet**  
A 10/100 network connection between the VoIP gateway and the Speech Server that supports VoIP.

**ETS**  
European Telecommunications Standard or European Telecommunication Specification.

**ETSI**  
European Telecommunications Standards Institute.

**Euro-ISDN**  
The common European ISDN standard, agreed in 1993, that provides a basic range of services and supplementary services using 30 B-channels plus a D-channel over an E1 trunk.

**exchange data link**  
A serial connection that carries messaging information between WebSphere Voice Response and the Lucent Technologies 1AESS, Northern Telecom DMS100, Ericsson MD110 switch, or Siemens Hicom 300.

**exit**  
A point in a supplied application from which control can be passed to another custom-written application. On completion, the custom-written application passes control back to the supplied application.

**F**

**fade in**  
To gradually increase the volume of sounds, such as background music.

**fade out**  
To gradually decrease the volume of sounds, such as background music.

**failover**  
A transparent operation that, in the event of a system failure, switches responsibility for managing resources to a redundant or standby system. Also known as failover.

**FDM**  
See Feature Download Management.

**Feature Download Management (FDM)**  
An ADSI protocol that enables several alternative key and screen overlays to be stored in an ADSI telephone, and to be selected by predetermined events at the telephone.
Federal Communication Commission (FCC)  The standard body in the United States that is responsible for communication.

field  An identifiable area in a window that is used to enter or display data.

FILELIST  A WebSphere Voice Server Telephony runtime file that defines which files to load into a WebSphere Voice Server engine. It contains a list in the form:

context type grammar filename
... ...

Recursion is not permitted; that is, no contexts of type FILELIST can be specified in a FILELIST. When a FILELIST is loaded, all the grammars that are specified in it are loaded into the engine. From then on, the grammars that are loaded when the FILELIST is specified are regarded as a single context.

Foreign Exchange Subscriber (FXS)  A signaling protocol that links a user's location to a remote exchange that would not normally be serving that user, to provide, for example, calls to outside the local area at the local rate.

frame  A group of data bits that is surrounded by a beginning sequence and an ending sequence.

fsg  Abbreviation for finite state grammar. In WebSphere Voice Server, the extension of a file that contains grammar specifications in compiled, binary form. It is generated from a .bnf file and is called a .fsg file.

function  In ADSI, an ADSI instruction or group of instructions.

FXS  See Foreign Exchange Subscriber

G
gatekeeper  A component of a Voice over Internet Protocol that provides services such as admission to the network and address translation.

gateway  A component of Voice over Internet Protocol that provides a bridge between VoIP and circuit-switched environments.

G.711  Specification for uncompressed voice for PSTN and Voice over Internet Protocol access.

G.723.1  Compressed audio codecs that are used on Voice over Internet Protocol connection for voice.

G.729A  Compressed audio codecs that are used on Voice over Internet Protocol connection for voice.

glare  A condition that occurs when both ends of a telephone line or trunk are seized at the same time.

grammar  A structured collection of words and phrases that are bound together by rules. A grammar defines the set of all words, phrases, and sentences that might be spoken by a caller and are recognized by the engine. A grammar differs from a vocabulary in that it provides rules that govern the sequence in which words and phrases can be joined together.
greeting
In voice mail, the recording that is heard by a caller on reaching subscriber's mailbox. See also announcement-only greeting. Contrast with voice message.

greeting header
In voice mail, a recording that is made by a subscriber and played to caller either before or instead of a personal greeting.

Groupe Special Mobile (GSM)
A CEPT/CCH standard for mobile telephony.

H
HACMP (High-Availability Cluster Multi-Processing) for AIX
Licensed Program Product (LPP) that provides custom software that recognizes changes in a cluster and coordinates the use of AIX features to create a highly-available environment for critical data and applications.

HACMP/ES
Licensed Program Product (LPP) that provides Enhanced Scalability to the HACMP for AIX LPP. An HACMP/ES cluster can include up to 32 nodes.

hang up
To end a call. See also disconnect.

HDB3
High-density bipolar of order 3. An E1 line coding method in which each block of four successive zeros is replaced by 000V or B00V, so that the number of B pulses between consecutive V pulses is odd. Therefore, successive V pulses are of alternate polarity so that no dc component is introduced. Note: B represents an inserted pulse that observes the alternate mark inversion (AMI) rule and V represents an AMI violation. HDB3 is similar to B8ZS that is used with T1.

HDLC
See high-level data link control.

high-level data link control
An X.25 protocol.

homologation
The process of getting a telephony product approved and certified by a country's telecommunications authority.

hook flash
A signal that is sent to a switch to request a switch feature (such as call transfer).

host application
An application residing on the host computer.

hunt group
A set of telephone lines from which a non-busy line is found to handle, for example, an incoming call.

I
immediate start
A procedure that is used with some channel-associated signaling protocols, when the address signaling is sent within 65 milliseconds of going off-hook. Contrast with delay start and wink start.

IN
See intelligent network.

in-band
In the telephony voice channel, signals are said to be carried in-band. Contrast with out-of-band.

indirect speech recognition
Identification of words from spoken
input that are read from a file. Contrast with direct speech recognition.

**initialize**
To prepare a system, device, or program for operation; for example, to initialize a diskette.

**input parameter**
Data that is received by a program such as a prompt, 3270 script, custom server, or state table from the program that called it. Contrast with local variable and system variable.

**integrated messaging**
A messaging system in which more than one copy of a single message is stored, the copies being kept synchronized by the applications that are used to access them. Contrast with unified messaging.

**Integrated Services Digital Network (ISDN)**
A digital end-to-end telecommunication network that supports multiple services including, but not limited to, voice and data.

**Integrated Services Digital Network (ISDN) call transfer**
In WebSphere Voice Response, an application that allows you to transfer calls on Nortel DMS-100 switches using Integrated Services Digital Network (ISDN) two B-channel transfer and on Nortel DMS-100 and DMS-250 switches using Nortel’s proprietary Release Link Trunk (RLT) call transfer protocol.

**Integrated Services Digital Network (ISDN) two B-channel transfer**
A call transfer feature that is defined by Bellcore GR-2865-CORE specification, and used on Nortel and Lucent switches.

**Integrated Services Digital Network user part (ISUP)**
Part of the SS7 protocol that supports telephony signaling applications. The ISDN user part is defined to carry signaling information that relates to digital telephones, terminals, and PABXs in customer premises.

**intelligent network (IN)**
A telephone network that includes programmable software that is not resident on the switch. It allows the service provider to provide special services, such as special call-handling, that are not dependent on the capabilities of the switch. See also advanced intelligent network.

**intelligent peripheral (IP)**
A voice processing system (such as WebSphere Voice Response) that provides enhanced services such as voice response, speech recognition, text-to-speech, voice messaging, and database access in an advanced intelligent network.

**interactive voice response (IVR)**
A computer application that communicates information and interacts with the caller via the telephone voice channel.

**International Telecommunications Union – Telecommunication Standardization Sector (ITU-T)**
The name of the organization that was previously known as the CCITT.

**IP**
See intelligent peripheral.
ISDN  See Integrated Services Digital Network (ISDN).

ISDN two B-channel transfer  See Integrated Services Digital Network (ISDN) two B-channel transfer.

ISDN-UP  See Integrated Services Digital Network user part.

ISUP  See Integrated Services Digital Network user part.

ITU-T  See International Telecommunications Union – Telecommunication Standardization Sector.

IVR  See interactive voice response.

J

Java Bean  A reusable Java component. See beans.

jump out  See call transfer.

K

key  (1) One of the pushbuttons on the telephone handset; sometimes referred to as a DTMF key. (2) A component of the keyboard that is attached to the computer system.

key pad  The part of the telephone that contains the pushbutton keys.

key pad mapping  The process of assigning special alphanumeric characters to the keys that are on a telephone key pad, so that the telephone can be used as a computer-terminal keyboard.

L

LAN  See local area network.

language model  For speech recognition, a set of acoustic shapes (in binary format) for a given set of words, in which word-to-word differences are maximized, but speaker-to-speaker differences are minimized. See also vocabulary.

LAPD  See link access protocol for the D-channel.

licensed program product (LPP)  A separately-priced program and its associated materials that bear an IBM copyright and are offered under the terms and conditions of a licensing agreement.

license server  A machine on a network that holds licenses and distributes them on request to other machines on the network.

line error  An error on the telephone line that causes the signal to be impaired.

link access protocol for the D-channel  An HDLC protocol used in ISDN that ensures a reliable connection between the network and the user. Often used as another name for Q.921.

local area network (LAN)  A network in which computers are connected to one another in a limited geographical area. WebSphere Voice Response communication with WebSphere Voice Server speech recognition, text-to-speech, and single system image (SSI) requires a LAN that is dedicated to that purpose (unless
both are installed on the same system. A token-ring network is a type of LAN.

**local variable**
A user-defined temporary variable that can be accessed only by the program (state table, prompt, or 3270 script) for which it is defined. Contrast with input parameter, system variable.

**M**

macro See system prompt

MAP See mobile application part

MB See megabyte

megabyte
(1) For processor storage and real and virtual memory, 1 048 576 bytes. (2) For disk storage capacity and transmission rates, 1 000 000 bytes.

Message Center
See Unified Messaging

message delivery preference
The subscriber's choice of whether voice mail is stored as voice mail only, as e-mail only, or as both voice mail and e-mail.

message delivery type
The format in which a voice message is delivered.

message signal unit (MSU)
An MTP packet that contains data.

message transfer part (MTP)
Part of the SS7 protocol that is normally used to provide a connectionless service that is roughly similar to levels one through three of the OSI reference model.

**message waiting indicator (MWI)**
A visible or audible indication (such as a light or a stutter tone) that a voice message is waiting to be retrieved.

MFR1 An in-band address signaling system that uses six tone frequencies, two at a time. MFR1 is used principally in North America and is described in ITU-T recommendations Q.310 through Q.332.

MIME See multipurpose Internet mail extensions.

mobile application part (MAP)
Optional layer 7 application for SS7 that runs on top of TCAP for use with mobile network applications.

MP See multiprocessor

MSU See message signal unit

MTP See message transfer part

mu(µ)-law
The companding algorithm that is used primarily in North America and Japan when converting from analog to digital speech data. (Comand is a contraction of compress and expand.) Contrast with A-law.

multiprocessor (MP)
A computer that includes two or more processing units that can access a common main storage.

multipurpose Internet mail extensions (MIME)
A protocol that is used on Internet for extending e-mail capability and merging it with other forms of communication, such as voice mail and fax.
mumble
Non speech noise that a user interjects while speaking.

music channel
A channel on which sounds can be broadcast to one or more telephony (voice) channels.

music title
The name by which WebSphere Voice Response knows a tune.

MWI  See message waiting indicator

N
National ISDN
A common ISDN standard that was developed for use in the U.S.

NAU  See network addressable unit

N-Best
The ability to return more than one speech recognition result. Typically, an array of results is available in the application in sequence of descending probability.

NCP  See network control program

NET Norme Européenne de Télécommunication.

Net 5 The test specification for conformance to the Euro-ISDN standard for primary rate access to ISDN.

network addressable unit (NAU)
Any network component that can be addressed separately by other members of the network.

network control program (NCP)
Used for requests and responses that are exchanged between physical units in a network for data flow control.

Network File System (NFS)
A protocol, developed by Sun Microsystems, Incorporated, that allows any host in a network to gain access to another host or netgroup and their file directories. In a single system image (SSI), NFS is used to attach the WebSphere Voice Response DB2 database.

general termination
See NT mode

NFAS See non-facility associated signaling

NFS See Network File System

node
In a single system image (SSI), one of the WebSphere Voice Response systems that are in the cluster.

non-facility associated signaling (NFAS)
An ISDN configuration where several T1 facilities can be controlled by a single D-channel, instead of the normal T1 configuration where each T1 facility has 23 B-channels and a D-channel (23B+D). With NFAS, all 24 timeslots of the non signaling trunks are available for voice, whereas only 23 channels can be used on the trunk that carries signaling traffic (23B+D+n24B).

NT mode
Attachment to the ISDN network is asymmetric. The network side of the connection operates in network termination, or NT, mode. User equipment operates in terminal equipment, or TE, mode.

O

ODM See Object Data Manager

Object Data Manager (ODM)
A data manager intended for the storage of system data. The ODM is
used for many system management functions. Information that is used in many commands and SMIT functions is stored and maintained in the ODM as objects with associated characteristics.

off-hook
A telephone line state, usually induced by lifting a receiver, in which the line is ready to make a call.

offline
Not attached or known to the existing system configuration, and therefore not in active operation.

on-hook
A telephone line state, usually induced by hanging up a receiver, in which the line is ready to receive a call.

online
In active operation.

OPC See originating point code.

Open Systems Interconnection (OSI)
(1.) The interconnection of open systems as specified in particular ISO standards. (2.) The use of standardized procedures to enable the interconnection of data processing systems.

Open Systems Interconnection (OSI) architecture
Network architecture that observes the particular set of ISO standards that relate to Open Systems Interconnection.

Open Systems Interconnection (OSI) Reference Model
A conceptual model composed of seven layers, each specifying particular network functions. Developed by the International Organization for Standardization (ISO) in 1984, it is considered to be the primary architectural model for intercomputer communications.

originating point code (OPC)
A code that identifies the signaling Point that originated an MTP signal unit. Unique in a particular network.

OSI See Open Systems Interconnection.

outgoing mail
In voice mail, messages that are sent by a subscriber to another subscriber on the same system, and have not yet been listened to by the addressee.

out-of-band
In the telephony signaling channel, as opposed to the voice channel. Signals are said to be carried out-of-band. Contrast with in-band.

P
PABX See private automatic branch exchange.

Pack
Each DTTA contains the equivalent of four packs. The pack is a digital trunk processor built into the digital trunk adapter, so there is no need for external hardware. See also TPACK.

Parameter file
An ASCII file that sets configuration parameters.

Password
A unique string of characters that is known to a computer system and to a user. The user must specify the character string to gain access to the system and to the information that is stored in it.

PBX See private branch exchange.
PCI  See peripheral component interconnect.

PCM  See Pulse Code Modulation.

PCM fault condition
A fault, such as power supply failure, or loss of incoming signal, in T1 or E1 equipment. (ITU-T G.732 and G.733.)

Peripheral component interconnect (PCI)
A computer busing architecture that defines electrical and physical standards for electronic interconnection.

Personal greeting
In voice mail, a greeting that is recorded by a subscriber. Contrast with system greeting.

Phone recognition
Communicating with a computer using voice via a telephone, over a telephone line. The computer application recognizes what was said and takes suitable action.

Port
In time-slot management, one end of a 64 Kbps unidirectional stream that can be attached to the TDM bus.

Port set
In time-slot management, a collection of ports that can be connected using a single CA_TDM_Connect() API call to a complementary collection of ports.

PRA  Primary rate access (PRA). Used as another name for primary rate interface (PRI).

PRI  See primary rate interface.

Primary rate access (PRA)
See primary rate interface.

Primary rate interface (PRI)
The means of ISDN access that is normally used by large sites. It provides 30 (E1) or 23 (T1) B-channels of 64 Kb per second and one D-channel for signaling. This is often known as 30B+D or 23B+D. Contrast with basic rate interface.

Primary rate ISDN (PRI)
See primary rate interface.

Primitive
A message that is sent from one process to another.

Private automatic branch exchange (PABX)
An automatic private switching system that services an organization and is usually located on a customer’s premises. Often used as another name for private branch exchange (PBX).

Private branch exchange (PBX)
A switch inside a private business that concentrates the number of inside lines into a smaller number of outside lines (trunks). Many PBXs also provide advanced voice and data communication features. Often used as another name for private automatic branch exchange.

Process a call
To answer the telephone and perform the correct tasks.

Process Manager
In WebSphere Voice Server, the process that manages the interaction of all telephony system processes; for example, starting and stopping text-to-speech or speech recognition sessions.

Production system
A WebSphere Voice Response system that responds to or makes “live” calls. A production system can also be used to develop new
program temporary fix (PTF)
An update to IBM software.

program data
Application-specific data that can be associated with a call transfer from CallPath to WebSphere Voice Response, or in the opposite direction. This is equivalent to CallPath program data, but WebSphere Voice Response imposes the restriction that the data must be a printable ASCII character string, with a maximum length of 512 bytes.

prompt
(1) A message that requests input or provides information. Prompts are seen on the computer display screen and heard over the telephone. (2) In WebSphere Voice Response, a program that uses logic to determine dynamically the voice segments that are to be played as a voice prompt.

prompt directory
A list of all the prompts that are used in a particular voice application. Used by the state table to play the requested voice prompts.

pronunciation
The possible phonetic representations of a word. A word can have multiple pronunciations; for example, “the” has at least two pronunciations, “thee” and “thuh”.

pronunciation dictionary
A file that contains the phonetic representation of all of the words, phrases, and sentences for an application grammar.

pronunciation pool
A WebSphere Voice Server resource that contains the set of all pronunciations.

protocol
A set of semantic and syntactic rules that determines the behavior of functional units when they get communication. Examples of WebSphere Voice Response protocols are FXS, RE, and R2.

PSTN
An ITU-T abbreviation for public switched telephone network.

PTF
See program temporary fix.

Pulse Code Modulation (PCM)
Variation of a digital signal to represent information.

pushbutton
(1) A key that is on a telephone key pad. (2) A component in a window that allows the user to start a specific action.

directory
A type of telephone that has pushbuttons. It might or might not send tone signals. If it does, each number and symbol on the key pad has its own specific tone.

Q

Q.921
The ITU-T (formerly CCITT) recommendation that defines the link layer of the DSS1 protocol. Q.921 defines an HDLC protocol that ensures a reliable connection between the network and the user. Often used as another name for LAPD.

Q.931
The ITU-T recommendation that defines the network layer of the DSS1 protocol. This layer carries the
ISDN messages that control the making and clearing of calls.

**quiesce**
To shut down a channel, a trunk line, or the whole system after allowing normal completion of any active operations. The shutdown is performed channel-by-channel. Channels that are in an idle state are shut down immediately. Channels that are processing calls are shut down at call completion.

R

RAI See remote alarm indication
RBS See robbed-bit signaling
RE See remote extension

**Recognition Engine server**
In WebSphere Voice Server, the software that performs the speech recognition and sends the results to the client. This consists of one ‘Tsm router’ and at least one ‘tsmp’ and one ‘engine’.

reduced instruction set computer (RISC)
A computer that uses a small, simplified set of frequently-used instructions to improve processing speed.

referral number
The phone number to which calls are routed, when call forwarding is active.

rejection
The identification of an utterance as one that is not allowed by a grammar.

release link trunk (RLT)
A custom specification from Nortel for ISDN call transfer.

**remote alarm indication (RAI)**
A remote alarm (also referred to as a yellow alarm) indicates that the far-end of a T1 connection has lost frame synchronization. The Send RAI system parameter can be set to prevent WebSphere Voice Response from sending RAI.

**remote extension (RE)**
An E1 signaling protocol that is similar to FXS loop start.

**resource element**
A component of an Intelligent Network. The resource element contains specialized resources such as speech recognizers or text-to-speech converters.

**response**
In speech recognition, the character string that is returned by the recognizer, through DVT_Client, to the state table. The string represents the result of a recognition attempt. This is the word or words that the recognizer considers to be the best match with the speech input.

result
An indicator of the success or failure of a state table action. It is returned by WebSphere Voice Response to the state table. Also known as an edge.

result state
The state that follows each of the possible results of an action.

return code
A code that indicates the status of an application action when it completes.

RISC See reduced instruction set computer
RLT See release link trunk
robbed-bit signaling (RBS)
The T1 channel-associated signaling scheme that uses the least significant bit (bit 8) of each information channel byte for signaling every sixth frame. This is known as 7-5/6-bit coding rather than 8-bit coding. The signaling bit in each channel is associated only with the channel in which it is contained.

S
SAP See service access point
SAS A T1 signaling protocol that is similar to FXS.
SCbus See Signal Computing bus
SCCP See signaling connection control part
SCP See service control point

screened transfer
A type of call transfer in which the transfer of the held party to the third party is completed only if the third party answers the call. Contrast with blind transfer.

script
The logical flow of actions for a 3270 server program.

script language
A high-level, application-specific scripting language, which consists of statements that are used to develop 3270 scripts. These scripts are part of the interface between a state table and a 3270-based host business application.

segment ID number
One or more numbers that are used to identify a voice or prompt segment.

Server Display Control (SDC)
An ADSI control mode in which the ADSI telephone is controlled through a dialog with a voice response system.

server node
In a single system image (SSI), a WebSphere Voice Response system that contains either the WebSphere Voice Response DB2 database, or the voice data, or both.

service access point (SAP)
An OSI term for the port through which a service user (layer N+1) accesses the services of a service provider (layer N).

service control point (SCP)
A component of the intelligent network that provides transactional services, such as translation of toll-free numbers to subscriber numbers.

service information octet (SIO)
A field that is in an MTP message signal unit. It identifies a higher layer user of MTP, and whether the message relates to a national or international network.

service node
An element of an Intelligent Network. The service node contains the service logic that controls an intelligent network application and resources.

service provider
Any company that provides services for a fee to its customers, such as telecommunication companies,
application service providers, enterprise IT, and Internet service providers.

**service provider equipment (SPE)**
The switching equipment that is owned by the telephone company.

**session** See [speech recognition session](#).

**Session Initiation Protocol**
A signaling protocol used for internet conferencing, telephony, presence, events notification and instant messaging.

**short message service center (SMSC)**
A component of the mobile telephony network, specified by the GSM group of standards, that provides for exchange of alphanumeric messages of less than 160 bytes. Messages can be exchanged between different types of system such as mobile telephone, alphanumeric pager, terminal, e-mail, telex, or DTMF telephone.

**SIF** See [signaling information field](#).

**Signal Computing System Architecture (SCSA)**
An architecture that was defined by Dialogic to support interoperability of software and hardware components that are developed by different vendors in the computer telephony industry.

**Signal Computing bus (SCbus)**
A time division multiplexed (TDM) hardware bus that was originated by Dialogic to interconnect different vendors’ computer telephony adapters. Specified as part of [Signal Computing System Architecture](#) (SCSA).

**signaling**
The exchange of control information between functional parts of the system in a telecommunications network.

**signaling connection control part (SCCP)**
A layer 3 protocol that observes OSI.

**signaling information field (SIF)**
The user data portion of an MTP message signal unit.

**signaling link code (SLC)**
A code that identifies a particular signaling link that connects the destination and originating signaling points. This is used in MTP signaling network management messages to indicate the signaling link to which the message relates.

**signaling link selection (SLS)**
A field that is used to distribute MTP signal units across multiple signaling links.

**signaling mode**
The type of signaling protocol, either channel-associated signaling, or common-channel signaling.

**signaling point**
A node in a signaling network that either originates and receives signaling messages, or transfers signaling messages from one signaling link to another, or both.

**signaling process**
A WebSphere Voice Response component that controls signaling for an exchange data link or common-channel signaling protocol. Some signaling processes are
supplied with WebSphere Voice Response, and others can be custom-written.

signaling System Number 7 (SS7)
The international high-speed signaling backbone used for the public-switched telephone network.

silence
A short pause between utterances.

simple mail transfer protocol (SMTP)
An Ethernet protocol that is related to TCP/IP.

simple network management protocol (SNMP)
In the Internet suite of protocols, a network management protocol that is used to monitor routers and attached networks. SNMP is an application layer protocol. Information on devices managed is defined and stored in the application’s Management Information Base (MIB). SNMP provides a means of monitoring WebSphere Voice Response resources remotely.

Simplified Message Desk Interface (SMDI)
A Northern Telecom service that transmits out-of-band information between WebSphere Voice Response and particular switches.

Simplified Message Service Interface (SMSI)
A Lucent Technologies service that transmits out-of-band information between WebSphere Voice Response and particular switches.

single system image (SSI)
A cluster of WebSphere Voice Response systems that are connected together using a local area network. Each system (known as a node) in the cluster is configured as either a client or a server. A single system image typically consists of one server node and multiple client nodes. The client nodes retrieve applications and voice data from the server. A second server can be configured for redundancy.

sink
A port that takes voice data from the TDM bus. Contrast with source.

SIO See service information octet.
SIP See Session Initiation Protocol.
SLC See signaling link code.
SLS See signaling link selection.
SMDI See Simplified Message Desk Interface.
SMIT See System Management Interface Tool.
SMP See symmetric multiprocessor.
SMSC See short message service center.
SMSI See Simplified Message Service Interface.
SMTP See simple mail transfer protocol.
SNA Systems Network Architecture.
SNMP See simple network management protocol.

source
A port that puts voice data on to the TDM bus. Contrast with sink.

SPACK
A logical component that consists of a base card, which connects to the digital trunk adapter in the pSeries computer, and a trunk interface card (TIC), which manages the trunk connection to the switch. Contrast with VPACK and TPACK.

SPE See service provider equipment.
**speaker-dependent speech recognition**
Identification of spoken words that is related to knowledge of the speech characteristics of one speaker. Contrast with **speaker-independent speech recognition**.

**speaker-independent speech recognition**
Identification of spoken words that is related to collected knowledge of the speech characteristics of a population of speakers. Contrast with **speaker-dependent speech recognition**.

**special character**
A character that is not alphabetic, numeric, or blank. For example, a comma (,) or an asterisk (*)

**speech recognition**
The process of identifying spoken words. See **discrete word recognition**, **continuous speech recognition**, **speaker-dependent speech recognition**, **speaker-independent speech recognition**

**Speech Recognition Control Language (SRCL)**
In WebSphere Voice Server, a structured syntax and notation that defines speech grammars, annotations, repetitions, words, phrases, and associated rules.

**speech recognition session**
In WebSphere Voice Server, a sequence of recognition commands that allocate a recognition engine, and return a unique identifier to identify the engine.

**speech synthesis**
The creation of an approximation to human speech by a computer that concatenates basic speech parts together. See also **text-to-speech**

**SRCL**
See **Speech Recognition Control Language (SRCL)**

**SS7**
See **signaling System Number 7**

**SSI**
See **single system image**

**SSI-compliant custom server**
A custom server that runs correctly in a single system image. The custom server observes all the guidelines for the operation of custom servers in an SSI environment.

**SSI-tolerant custom server**
A custom server that runs in a single system image, but with only some restrictions.

**standalone system**
A WebSphere Voice Response system that is not part of a single system image (SSI). A standalone system is not connected to other WebSphere Voice Response systems, so it contains its own application and voice data.

**state**
One step in the logical sequence of actions that makes a WebSphere Voice Response voice application.

**state table**
A list of all the actions that are used in a particular voice application. A component of WebSphere Voice Response.

**state table action**
One instruction in a set of instructions that is in a WebSphere Voice Response state table that controls how WebSphere Voice Response processes various operations such as playing voice prompts or recording voice messages. See also **state**
stub  A line in a state table that is only partially displayed.

subscriber  In voice mail, any person who owns a mailbox.

subscriber class  A named set of variables that defines a specific level of service available to telephone subscribers, such as maximum number of messages per mailbox and maximum number of members per mailbox distribution list.

subvocabulary  A vocabulary that is called by another vocabulary.

supplementary service  In Euro-ISDN, a service outside the minimum service offering that each signatory is obliged to provide. For example, calling line identification presentation (CLIP) and call session.

switch  A generic term that describes a telecommunications system that provides connections between telephone lines and trunks.

symmetric multiprocessor (SMP)  A system in which functionally-identical multiple processors are used in parallel, providing simple and efficient load-balancing.

Synchronous Data Link Control (SDLC)  A discipline for managing synchronous, code-transparent, serial-by-bit information transfer over a link connection. Transmission exchanges can be duplex or half-duplex over switched or nonswitched links.

system administrator  The person who controls and manages the WebSphere Voice Response system by adding users, assigning account numbers, and changing authorizations.

system greeting  In voice mail, a default greeting that is heard by callers to the mailboxes of subscribers who have not recorded a personal greeting or who have selected the system greeting. Contrast with personal greeting.

System Management Interface Tool (SMIT)  A set of utilities that can be used for various purposes, such as loading WebSphere Voice Response software, installing the exchange data link, and configuring SNA.

Systems Network Architecture (SNA)  An architecture that describes the logical structure, formats, protocols, and operational sequences for transmitting information units through the networks and also the operational sequences for controlling the configuration and operation of networks.

system parameter  A variable that controls some of the behavior of WebSphere Voice Response or applications that are running under WebSphere Voice Response. System parameters are set through System Configuration or Pack Configuration options on the Configuration menu. Some system parameter values are assigned to system variables when an application is initialized. Contrast with input parameter, local variable, system variable.

system prompt  The symbol that appears at the
command line of an operating system, indicating that the operating system is ready for the user to enter a command.

**system variable**
A permanent global variable that is defined by WebSphere Voice Response for use by state tables. Many system variables are loaded with values when the state table is initialized. Some values are taken from system parameters. Contrast with input parameter, local variable, system parameter.

**T**

**T1** A digital trunking facility standard that is used in the United States and elsewhere. It can transmit and receive 24 digitized voice or data channels. Signaling can be imbedded in the voice channel transmission when robbed-bit signaling is used. The transmission rate is 1544 kilobits per second. Contrast with T1/D3 or T1/D4.

**T1/D3** A framing format that is used in T1 transmission.

**T1/D4** A framing format that is used in T1 transmission.

**tag** A text string that is attached to any instance of a word in a grammar. A tag can be used (1) to distinguish two occurrences of the same word in a grammar or (2) to identify more than one word in a grammar as having the same meaning.

**Tag Image File Format-Fax (TIFF-F)** A graphic file format that is used to store and exchange scanned fax images.

**TCAP** See [transaction capabilities application part](#).

**TCP/IP** See [Transmission Control Protocol/Internet Protocol](#).

**TDD** See [Telecommunications Device for the Deaf](#).

**TDM** See [time-division multiplex bus](#).

**technology** A program, external to WebSphere Voice Response, that provides processing for functions such as text-to-speech or speech recognition.

**Telecommunications Device for the Deaf (TDD)** A telephony device that has a QWERTY keyboard and a small display and, optionally, a printer.

**telephone input field** A field type that contains information that is entered by a caller who is using pushbutton signals. See also field.

**terminal** (1) A point in a system or communication network at which data can enter or leave. (2) In data communication, a device, usually equipped with a keyboard and display device, that can send and receive information.

**termination character** A character that defines the end of a telephone data entry.

**text-to-speech (TTS)** The process by which ASCII text data is converted into synthesized speech. See also [speech synthesis](#).

**TIC** See [trunk interface card](#).

**time-division multiplex bus (TDM)** A method of transmitting many
channels of data over a smaller number of physical connections by multiplexing the data into timeslots, and demultiplexing at the receiving end. In this document, one such channel can be considered to be a half-duplex unidirectional stream of 64 Kb per second.

TIFF-F
See Tag Image File Format-Fax

timeslot
The smallest switchable data unit on a data bus. It consists of eight consecutive bits of data. One timeslot is similar to a data path with a bandwidth of 64 Kb per second.

token
A particular message or bit pattern that indicates permission or temporary control to transmit.

token-ring network
A local area network that connects devices in a ring topology and allows unidirectional data transmission between devices by a token-passing procedure. A device must receive a token before it can transmit data.

tone
An audible signal that is sent across a telephone network. Single (one-frequency) tones, tritones (three sequential tones at different frequencies), dual tones (two simultaneous tones at different frequencies), and dual sequential tones exist. Each has a different meaning.

TPACK
A digital trunk processor that is implemented using DSP technology on the digital trunk adapter without the need for external hardware. One DTTA digital trunk adapter provides up to four TPACKs on a PCI card.

transaction
A specific, related set of tasks in an application that retrieve information from a file or database. For example, a request for the account balance or the available credit limit.

transaction capabilities application part (TCAP)
Part of the SS7 protocol that provides transactions in the signaling network. A typical use of TCAP is to verify a card number, for the credit card calling service.

transaction messaging
The ability to associate an item of data, such as a transaction identifier, with a voice message. The voice message can later be retrieved by referencing the data value.

transfer
See call transfer.

Transmission Control Protocol/Internet Protocol (TCP/IP)
A communication subsystem that is used to create local area and wide area networks.

trombone
A connected voice path that enters an IVR from a switch on one circuit, then returns to the same switch on a parallel circuit. Two IVR ports and two circuits are consumed, but in some circumstances this might be the only way to make a connection between two callers if the attached switch does not support a Call Transfer function. Also known as double-trunking.

trunk
A telephone connection between
two central offices or switching devices. In WebSphere Voice Response, a trunk refers to 24 or 30 channels that are carried on the same T1 or E1 digital interface.

**trunk interface card (TIC)**
The component of the pack that manages the trunk connection to the switch.

**Tsm Router**
In WebSphere Voice Server, a process that controls which engine processes are in use at any time. Requests for an engine by a WebSphere Voice Server Client are accepted or rejected depending on whether an engine that meets the Tsm Client's requirements is available.

**ttmp**
In WebSphere Voice Server, a process that is running on the Recognition engine server machine that passes messages between an engine and a Tsm Client. One ttmp exists for every engine.

**TTS**
See text-to-speech.

**tune**
A piece of music or other audio data that is intended to be played as background music.

**U**

**unerrun**
To run out of audio data to play, causing voice or music to be audibly broken up or cut off.

**unified messaging**
A messaging system in which a single copy of a message is stored and accessed by multiple applications (for example, voice mail and e-mail). Contrast with integrated messaging.

**Unified Messaging**
An IBM product that uses WebSphere Voice Response's voice processing capabilities to provide a wide range of voice mail, fax, and e-mail functions. Previously known as Message Center.

**user**
Someone who uses WebSphere Voice Response as a system administrator, application developer, or similar. Contrast with caller.

**utterance**
A spoken word, phrase, or sentence that can be preceded and followed by silence.

**V**

**variable**
A system or user-defined element that contains data values that are used by WebSphere Voice Response voice applications. See input parameter, local variable, system parameter, system variable.

**VMS**
See Voice Message Service.

**vocabulary**
A list of words with which WebSphere Voice Response matches input that is spoken by a caller. See also language model.

**voice application**
A WebSphere Voice Response application that answers or makes calls, plays recorded voice segments to callers, and responds to the caller's input.

**voice directory**
A list of voice segments that is identified by a group ID. Voice directories can be referenced by prompts and state tables. Contrast with voice table.
voice mail
The capability to record, play back, distribute, and route voice messages.

voice mailbox
The notional hard disk space where the incoming messages for a voice mail subscriber are stored.

voice message
In voice mail, a recording that is made by a caller for later retrieval by a subscriber.

Voice Message Service (VMS)
An Ericsson service that transmits information between WebSphere Voice Response and particular switches.

voice messaging
The capability to record, play back, distribute, route, and manage voice recordings of telephone calls through the use of a processor, without the intervention of agents other than the callers and those who receive messages.

voice model
A file that contains parameters that describe the sounds of the language that are to be recognized on behalf of an application. In WebSphere Voice Server, this is a bnf file. See also grammar.

Voice over Internet Protocol (VoIP)
The sending of telephony voice over Internet Protocol (IP) data connections instead of over existing dedicated voice networks, switching and transmission equipment. See also gatekeeper and gateway.

voice port library
A library that manages a socket connection from the client to the voice technology. The library uses entry points that are provided by DVT.

Voice Protocol for Internet Messaging (VPIM)
The standard for digital exchange of voice messages between different voice mail systems, as defined in Internet Request For Comments (RFC) 1911.

voice response unit (VRU)
A telephony device that uses prerecorded voice responses to provide information in response to DTMF or voice input from a telephone caller.

voice segment
The spoken words or sounds that make recorded voice prompts. Each segment in an application is identified by a group ID and a segment ID and usually includes text.

voice server node
In a single system image (SSI), a server node that contains the voice data. This is usually the same node as the database server node.

voice table
A grouping of voice segments that is used for organizational purposes. Voice tables can be referenced by prompts, but not by state tables. Contrast with voice directory.

voice technology
See technology.

VoiceXML
VoiceXtensible Markup Language. An XML-based markup language for creating distributed voice applications. Refer to the VoiceXML forum web site at www.voicexml.org
VoIP  See *Voice over Internet Protocol*.

VPACK
A component consisting of a base card, which connects to the digital trunk adapter in the pSeries computer, and a trunk interface card (TIC), which manages the trunk connection to the switch. The single digital trunk processor contains one VPACK, and the multiple digital trunk processor contains slots for up to five VPACKs. Contrast with SPACK and TPACK.

VPIM  See *Voice Protocol for Internet Messaging*.

VRU  See *voice response unit*.

W

World Wide Web Consortium (W3C)
An organization that develops interoperable technologies (specifications, guidelines, software, and tools) to lead the Web to its full potential. W3C is a forum for information, commerce, communication, and collective understanding. Refer to the web site at http://www.w3.org

WebSphere Voice Response
A voice processing system, that combines telephone and data communications networks to use, directly from a telephone, information that is stored in databases.

wink start
A procedure that is used with some channel-associated signaling protocols to indicate when a switch or PABX is ready to accept address signaling. After seizure, the switch sends a short off-hook signal (wink) when it is ready to accept address information. Contrast with delay start and immediate start.

word spotting
In speech recognition, the ability to recognize a single word in a stream of words.

wrap
In ADSI, the concatenation of two columns of display data to form a single column.

Y

yellow alarm
See remote alarm indication.

Z

zero code suppression (ZCS)
A coding method that is used with alternate mark inversion to prevent sending eight successive zeros. If eight successive zeros occur, the second-least significant bit (bit 7, with the bits labeled 1 through 8 from the most significant to the least significant) is changed from a 0 to a 1. AMI with ZCS does not support clear channel operation.
List of WebSphere Voice Response and associated documentation

Here is a list of the documentation for WebSphere Voice Response for AIX and associated products. PDF and HTML versions of the documentation are available from the IBM Publications Center at http://www.ibm.com/shop/publications/order. Hardcopy books, where available, can be ordered through your IBM representative or at this Web site.

WebSphere Voice Response for AIX documentation can also be found by going to the IBM Pervasive software Web site at http://www.ibm.com/software/pervasive, selecting the WebSphere Voice products link, and then selecting the library link from the WebSphere Voice Response page.

PDF and HTML versions of the WebSphere Voice Response for AIX publications are available on the CD-ROM supplied with the product. In addition, WebSphere Voice Response for AIX, WebSphere Voice Response for Windows, Unified Messaging, and other WebSphere Voice publications are available together in PDF and HTML formats on a separately-orderable CD-ROM (order number SK2T-1787).

Note: To read PDF versions of books you need to have the Adobe Acrobat Reader (it can also be installed as a plug-in to a Web browser). It is available from Adobe Systems at http://www.adobe.com.

WebSphere Voice Response software

- WebSphere Voice Response for AIX: General Information and Planning, GC34-7084
- WebSphere Voice Response for AIX: Installation, GC34-7095
- WebSphere Voice Response for AIX: User Interface Guide, SC34-7091
- WebSphere Voice Response for AIX: Configuring the System, SC34-7078
- WebSphere Voice Response for AIX: Managing and Monitoring the System, SC34-7085
- WebSphere Voice Response for AIX: Designing and Managing State Table Applications, SC34-7081
- WebSphere Voice Response for AIX: Application Development using State Tables, SC34-7076
- WebSphere Voice Response for AIX: Developing Java applications, GC34-7082
WebSphere Voice Response for AIX: Deploying and Managing VoiceXML and Java Applications, GC34-7080
WebSphere Voice Response for AIX: Custom Servers, SC34-7079
WebSphere Voice Response for AIX: 3270 Servers, SC34-7075
WebSphere Voice Response for AIX: Problem Determination, GC34-7087
WebSphere Voice Response for AIX: Fax using Brooktrout, GC34-7083
WebSphere Voice Response for AIX: Cisco ICM Interface User’s Guide, SC34-7077
WebSphere Voice Response for AIX: MRCP for State Tables, SC34-7086
WebSphere Voice Response for AIX: Programming for the ADSI Feature, SC34-7088
WebSphere Voice Response for AIX: Programming for the Signaling Interface, SC34-7089
WebSphere Voice Response for AIX: Voice over IP using Session Initiation Protocol, GC34-7093
WebSphere Voice Response for AIX: Using the CCXML Browser, SC34-7092

IBM hardware for use with WebSphere Voice Response
- IBM Quad Digital Trunk Telephony PCI Adapter (DTTA): Installation and User’s Guide, part number 00P3119 (DTTA card)

WebSphere Voice Response related products

WebSphere Voice Server
The documentation for Version 5.1 of WebSphere Voice Server is provided in the form of an HTML-based information center, and can be found at:

Unified Messaging for WebSphere Voice Response
- Unified Messaging: General Information and Planning, GC34-6398
- Unified Messaging: Subscriber’s Guide (Types 0, 1, 2, 3, 4 and 9), SC34-6403
- Unified Messaging: Subscriber’s Guide (Types 5, 6, 7 and 8), SC34-6400
- Unified Messaging: Administrator’s Guide, SC34-6399
- Unified Messaging: Voice Interface, GC34-6401
- Unified Messaging: Web Services Voicemail API, SC34-6975

Unified Messaging publications can be found by going to the IBM Pervasive software Web site at http://www.ibm.com/software/pervasive, selecting the products link, and then selecting the library link from the Unified Messaging page.
AIX and the IBM pSeries computer

For information on AIX Version 6.1, refer to the [AIX V6.1 infocenter](#).

For information on System p5 and BladeCenter computers, refer to the [IBM Power hardware infocenter](#).

HACMP

- HACMP for AIX: HACMP 5.4 Concepts and Facilities, SC23-4864-09
- HACMP for AIX: HACMP 5.4 Planning Guide, SC23-4861-09
- HACMP for AIX: HACMP 5.4 Installation Guide, SC23-5209-00
- HACMP for AIX: HACMP 5.4 Administration Guide, SC23-4862-09
- HACMP for AIX: HACMP 5.4 Smart Assist for DB2, SC23-5179-03
- HACMP for AIX: HACMP 5.4 Troubleshooting, SC23-5177-03
- HACMP for AIX: Enhanced Scalability Installation and Administration Guide, Volume 1, SC23-4284

For more information on HACMP, refer to the [HACMP Library](#) and the [AIX V6.1 infocenter](#).

SS7


IBM SS7 Support for WebSphere Voice Response observes the applicable parts of the following specifications for ISUP:
  - ITU-T (formerly CCITT) Recommendations Q.700 - Q.716, Volume VI Fascicle VI.7
  - ITU-T (formerly CCITT) Recommendations Q.721 - Q.725, Volume VI Fascicle VI.8
  - ITU-T (formerly CCITT) Recommendations Q.771 - Q.775, Q.791, Volume VI Fascicle VI.9

ADC

**Integrated Services Digital Network**

WebSphere Voice Response ISDN support observes the applicable parts of the following standards for User Side protocol:

**Custom ISDN Standards:**
- *Northern Telecom DMS/250 Primary Rate Interface* NIS A211-4 Release 8, July 1995. (IEC05 level)
- *Northern Telecom DMS/100 Primary Rate Interface* NIS A211-1 Release 7.05, May 1998. (NA007 & RLT)
- *AT&T 5ESS Switch ISDN Primary Rate Interface Specification. 5E7 and 5E8 Software Release* AT&T 235-900-332. Issue 2.00 December 1991
- *AT&T 5ESS Switch ISDN Primary Rate Interface Specification. 5E9 Software Release* AT&T 235-900-342. Issue 1.00 November 1993 (National ISDN only)
- *Lucent 5ESS-2000 Switch ISDN Primary Rate Interface, Interface Specification, 5E9(2) and Later Software Releases, 235-900-342. Issue 5.00 January 1997* (National ISDN only)
- *AT&T ISDN Primary Rate Specification TR41449 July 1989*
- *AT&T ISDN Primary Rate Specification TR41459 August 1996*

**Euro-ISDN**

The following documents refer to the specifications required for observing ISDN:
- *TBR4-ISDN; Attachment Requirements For Terminal Equipment To Connect To An ISDN Using ISDN Primary Rate Access, Edition 1, Nov. 95, English*
- *CTR 4 - European Communities Commission Decision 94/796/EC published in the Official Journal of the European Communities L 329, 20 December 94 (ISDN PRA)*

**National ISDN**

National ISDN is described in the following publications:
- *National ISDN, SR-NWT-002006, Issue 1, August 1991, published by Bellcore*
- *National ISDN-2, SR-NWT-002120, Issue 1, May 1992, published by Bellcore*

**INS Net Service 1500**

INS Net Service is described in the following publications:
Bellcore Specifications for ADSI Telephones

The following Bellcore specification documents contain technical details of the requirements for ADSI telephones, and the interface to voice response systems such as WebSphere Voice Response:

- SR-INS-002461: CustomerPremises Equipment Compatibility Considerations for the Analog Display Services Interface
- TR-NWT-001273: Generic Requirements for an SPCS to Customer Premises Equipment Data Interface for Analog Display Services
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