WebSphere Voice Response for AIX with DirectTalk Technology

MRCP for State Tables

Version 6.1
WebSphere Voice Response for AIX with DirectTalk Technology

MRCP for State Tables

Version 6.1
Note

Before using this information and the product it supports, read the general information under “Notices” on page 27.
Figures

1. Example state table for an MRCP text-to-speech application . . . . . 9
About this information

This book describes the new text-to-speech support for IBM® WebSphere® Voice Response for AIX® with DirectTalk® Technology client state tables that allows WebSphere Voice Response to use voice synthesis (text-to-speech) servers supporting the Media Resource Control Protocol (MRCP), Version 1.0.

In conjunction with this book, you should read the MRCP V1 specification available at


The latest version of the MRCP V1 specification is available at

http://rfc.net/rfc4463.html

Who should use this information

This book is for developers or system administrators who want to develop or maintain IBM WebSphere Voice Response for AIX solutions that use text-to-speech.

System administrators should be familiar with:

- The operation of pSeries® or BladeCenter® computers running AIX
- WebSphere Voice Response
- TCP/IP networking
- Installing and configuring WebSphere Voice Response custom servers

To install MRCP for state tables, some experience with the AIX operating system and its system management interface tool (SMIT) would be useful. No particular knowledge is necessary for configuration, other than the ability to use a text editor.

To develop applications, familiarity with creating state tables for WebSphere Voice Response applications is an advantage. If you have not worked with WebSphere Voice Response applications before, you should first read the introduction in the WebSphere Voice Response for AIX: General Information and Planning.

A basic knowledge of how text-to-speech (TTS) applications work, and how to create them would also be useful.
How to use this information

This book is divided into two main parts:

- An introduction to MRCP and how to use it to create voice applications that use text-to-speech.
- A sequence of reference topics describing the MRCP for state tables application programming interface (API).

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 products such as IBM WebSphere Voice Server. 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 65).

**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 65.
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

If you especially like or dislike anything about this book, feel free to send us your comments.

You can comment on what you regard as specific errors or omissions, and on the accuracy, organization, subject matter, or completeness of this book. Please limit your comments to the information that is in this book and to the way in which the information is presented. Speak to your IBM representative if you have suggestions about the product itself.

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Please ensure that you include the book title, order number, and edition date.
Chapter 1. Using MRCP for state tables

This chapter gives a general overview of the Media Resource Control Protocol (MRCP) and how it is implemented for WebSphere Voice Response state tables.

What is the Media Resource Control Protocol (MRCP)?

The Media Resource Control Protocol (MRCP) was developed jointly by Cisco Systems Inc., Nuance Communications, and Speechworks Inc. to allow client devices such as personal computers or mobile phones to control 'media resources' such as speech synthesizers and speech recognizers, which typically would be configured to run on IP network-based servers.

MRCP defines the request, responses, and events necessary for controlling the media processing resources and uses a lower level protocol known as the Real Time Streaming Protocol (RTSP) as the means by which MRCP messages are communicated over the IP network. RTSP runs on top of the Transmission Control Protocol (TCP) in order to achieve reliable communication over Ethernet, for example. Note that the current release of WebSphere Voice Response MRCP for State Tables only supports Version 1 of MRCP (which uses RTSP as the lower layer protocol). As well as providing a transport layer for MRCP messages, the RTSP control session is also used to set up the characteristics for the Real Time Protocol (RTP) streams which are used to pass the actual voice data between the MRCP server and the client for Text To Speech. The WebSphere Voice Response MRCP for State Tables implementation conforms to the MRCP V1 RFC 4463 specification.

The WebSphere Voice Response MRCP state table API

The WebSphere Voice Response MRCP client state table API provides a means by which WebSphere Voice Response state table applications can access MRCP text-to-speech (TTS) servers attached to the WebSphere Voice Response client over an IP network. It consists of a set of state tables (each named with a prefix of MST_) which can be called directly from a State Table application. These state tables then call an IBM-supplied custom server (MST) which contains the MRCP client implementation.

Note that applications should always access MRCP using the supplied state tables and not by directly calling the MST custom server (which would lead to unpredictable results). A second custom server (MRCP) controls the RTP voice streaming and is common between the VoiceXML and State Table MRCP implementations.
The MRCP state table API supports a maximum of one text-to-speech (TTS) session per state table application instance. This means that one WebSphere Voice Response MRCP client can have up to 240 active sessions running simultaneously, and each session can have a single TTS service. This means that if, for example, multiple languages are to be supported, both cannot be active at the same time and that one session must be shut down before another can be used.

The API allows the application to specify MRCP header fields on certain of the API state tables. These are not checked by the API and are simply passed transparently through to MRC protocol. Therefore any errors in these headers will always be reported by the text-to-speech servers. When this happens, the API will return a protocol error return code with further details about the exact error being logged to the WebSphere Voice Response error log. For further information about exactly why the header is being rejected, you may need to refer to the server-side logs.

For details of all MRCP header fields, refer to the MRCP V1 specification available at [http://rfc.net/rfc4463.html](http://rfc.net/rfc4463.html). Application developers who want to trace MRCP message flow should take advantage of such facilities provided on the text-to-speech servers, this facility is not provided by the MRCP client state table API. Alternatively, the AIX iptrace utility can be used to trace all IP traffic on the WebSphere Voice Response client with the resulting binary file either being formatted using the AIX ipreport command, or alternatively it can be imported into a popular IP traffic analysis tool.
Chapter 2. Installing and configuring MRCP for State Tables

This chapter tells you how to install the MRCP for State Tables custom servers and also how to configure the MRCP client state table API.

Installing MRCP for State Tables

To install MRCP for State Tables:

1. If you have not already used the vae.setuser utility, or you have installed WebSphere Voice Response software since it was last used, you must run it before using the vaeinit command. Refer to the section “Setting the dtuser file permissions” in WebSphere Voice Response for AIX: Installation for instructions.

2. Log on to the AIX user account that is set up for WebSphere Voice Response (normally dtuser). This should start the initialization sequence and display the WebSphere Voice Response User 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)

   If the User Login menu is not displayed, type vaeinit and press Enter.

3. Type 1 and press Enter.

   The system prompts you for your display name.

4. Type your machine’s IP address followed by :0

5. When the system displays the Logon window, log in to WebSphere Voice Response as admin.
   a. If you already use MRCP for VoiceXML applications, check if any of the MST, MRCP or MRCP_Log custom servers are running. If so, use the Custom Server Manager to stop them.
   b. From a shell window type the following commands:

      cd $VAE/sw/samples/MST/
      MST_Import -i MST.tar.Z

      For more information, refer to the $VAE/sw/samples/MST/readme.txt file.

6. Check that no errors were generated. If there were no errors, you have just successfully installed the following custom servers:

   MST
Configuring MRCP for state tables

Configuration of the WebSphere Voice Response MRCP client state table API is defined in the mst.xml file within directory: $CUR_DIR/ca/MST_dir

This file contains XML definitions of specific engines (TTS resources) and the general configuration of the system. The file is processed when the MST custom server is started. This is also when the domain names specified are resolved to IP addresses. Any change made after MST has started will not be effective until after it has been restarted. If a domain name cannot be resolved on start up, the engine will not be added. A sample file, $CUR_DIR/ca/MST_dir/mst_sample.xml, is supplied for reference.

The mst.xml file contains mapping between engine names (for example, gbtts) and a specific MRCP server (for example, machine.hursley.ibm.com, port 554). The engine names specified in file mst.xml define a TTS resource on an MRCP server.

<?xml version="1.0"?>
<!-- This is the MRCP for State Tables config file -->
<config>
    <!-- engine definitions -->
    <engine name="gbtts" type="tts">
        <server>
            machine.hursley.ibm.com
        </server>
        <port>
            554
        </port>
        <mediaurl>
            media/synthesizer
        </mediaurl>
        <initparms>
            speech-language:en-gb
            voice-age:25
        </initparms>
    </engine>
</config>

Valid XML elements:

debug
    Defines debug options and supports the following elements:

level
    Tracing level of the system. This can either be a number or a predefined trace level string. Contact IBM Support for further information.
Components to trace into the AIX trace channel 1 (WVR trace).
Comma separated list of component names. Contact IBM Support for further information.

**engine**
- name attribute defines the engine name, type attribute defines tts

**server**
Address of the text-to-speech server (mandatory)

**port**
The port to use on the server (mandatory)

**mediaurl**
A relative URL specifying synthesis server location. This is appended to address for use with RTSP messages. (optional)

**initparms**
Header fields to be passed on the initial SET-PARMS MRCP message subsequent to the MRCP session being established (mandatory). This element must contain the speech-language: MRCP header field for use as the default language for the engine.

Table 1. Client to MRCP synthesizer header field parameters for use with WebSphere Voice Response. All header fields are optional.

<table>
<thead>
<tr>
<th>Synthesizer-header field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>jump-target</td>
<td>Position to jump forward or backwards in the text being played, relative to the current position.</td>
</tr>
<tr>
<td>speaker-profile</td>
<td>The URI of a configuration file that includes a set of voice parameters such as gender, accent, and so on.</td>
</tr>
<tr>
<td>voice-parameter</td>
<td>A set of one or more voice parameters such as gender, accent, and so on.</td>
</tr>
<tr>
<td>prosody-parameter</td>
<td>A set of one or more prosody parameters such as volume, speed, and so on.</td>
</tr>
<tr>
<td>vendor-specific</td>
<td>Allows vendor-specific attributes and values to be sent to the synthesizer to set vendor-specific parameters. Refer to the vendor’s documentation for information.</td>
</tr>
<tr>
<td>speech-marker</td>
<td>Marker tag to be inserted in the speech data so that an event can be generated at that point.</td>
</tr>
<tr>
<td>speech-language</td>
<td>The code for the language of the synthesizer, for example, en-US.</td>
</tr>
<tr>
<td>fetch-hint</td>
<td>URI access properties to be used when resources such as documents are retrieved.</td>
</tr>
<tr>
<td>audio-fetch-hint</td>
<td>URI access properties to be used when resources such as speech audio files are retrieved.</td>
</tr>
<tr>
<td>fetch-timeout</td>
<td>Specifies the timeout duration to allow for retrieving resources.</td>
</tr>
</tbody>
</table>
Table 1. Client to MRCP synthesizer header field parameters for use with WebSphere Voice Response (continued). All header fields are optional.

<table>
<thead>
<tr>
<th>Synthesizer-header field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>speak-length</td>
<td>The position in the text being played up to which speech is to be played, relative to the current position.</td>
</tr>
</tbody>
</table>

Refer to the MRCP V1 specification available at http://rfc.net/rfc4463.html for full details. These parameters can be overridden for a specific session by using the MST_TTS_Config on page 20 state table.

The kill-on-merge-in header field parameter is not supported by the WebSphere Voice Response MRCP state table API. Do not use it in the <initparms> configuration. For details of how to set the behavior for merge-in using DTMF, see the information for the stop_keys parameter of MST_TTS_Speak.
Chapter 3. Designing and creating an MRCP voice application

This chapter describes the various components of a WebSphere Voice Response state table application that uses speech synthesis and how to use them.

State table voice application components

To create a WebSphere Voice Response state table voice application that uses the MRCP API, you need to develop various components for the WebSphere Voice Response client system and the MRCP telephony Text-To-Speech system. Both these systems provide a range of easy-to-use tools to aid the process. The following chapters describe each component, and the tools used to develop them.

A WebSphere Voice Response state table application consists of:

- State tables, which control interaction with the caller and with MST and other custom servers
- Prompts, which control the voice segments played to the caller
- Voice segments, the spoken output heard by the caller
- An application profile, which connects the incoming call to a specific state table

and, optionally custom servers that give you access to application data.

WebSphere Voice Response AIX applications are written by creating state tables. The standard WebSphere Voice Response system actions allow the application to do such things as wait for, or make, a phone call, play voice.

The additional state tables supplied for MRCP extend the set of actions available to WebSphere Voice Response applications, enabling the applications to:

- Take control of and manage text-to-speech resources and use them to synthesize speech from text strings.

WebSphere Voice Response MRCP TTS applications

The following state tables enable you to include MRCP Text-To-Speech in voice applications for the WebSphere Voice Response telephony connection environment:

- "MST_TTS_Assign" on page 19
MST_TTS_Assign
Open a session with the Text-To-Speech server. See "MST_TTS_Assign" on page 19.

MST_TTS_Config
Optionally configures a session with the Text-To-Speech server, augmenting and updating the default configuration in file $CUR_DIR/ca/MST_dir/mst.xml. See "MST_TTS_Config" on page 20.

MST_TTS_Free
Closes a session with the Text-To-Speech server. See "MST_TTS_Free" on page 24.

MST_TTS_Speak
Synthesizes and plays a text string or text file. See "MST_TTS_Speak" on page 21.

These state tables are described fully in Chapter 4, “MRCP TTS state tables,” on page 19.

Designing the state table
This section tells you how to design an MRCP speech synthesis application and implement it using the supplied state tables:

- "Application flow" maps the steps to state tables.
- "Creating the state table" on page 9 explains how to create a state table for a simple MRCP text-to-speech application.

Application flow
Once you have designed your application, you can implement it in the form of state table:

The application flow for implementing Text-To-Speech in the form of state tables is as follows:

assign (connect to) a Text-To-Speech engine
AIX state table - "MST_TTS_Assign" on page 19
Text-To-Speech engine
optionally configure the Text-To-Speech engine
AIX state table - "MST_TTS_Config" on page 20
synthesize and play text one or more times
AIX state table - "MST_TTS_Speak" on page 21
free the Text-To-Speech engine
AIX state table - "MST_TTS_Free" on page 24
Creating the state table

To implement MRCP text-to-speech, you create a new state table using the WebSphere Voice Response State Table editor. Alternatively, you can create a state table using an ASCII editor and then import it to the WebSphere Voice Response State Table editor.

Figure 1 shows a completed state table for a simple MRCP text-to-speech application in a WebSphere Voice Response State Table editor window List View.

Note: Refer to the WebSphere Voice Response for AIX WebSphere Voice Response for AIX: Application Development using State Tables manual for general information on creating state tables.

To create a new state table application for MRCP text-to-speech using the WebSphere Voice Response State Table editor:

1. Create the state table:
   a. From the WebSphere Voice Response Welcome window, select Applications → State Tables to display the State Tables window.
   b. Click File → New. An empty State Table window is displayed:
c. Click View → List View to display the state table as a list, as shown in the figures in this chapter.

2. Define any entry points, input parameters, and local variables that you intend to use in your state table. For MRCP text-to-speech you need to specify:
   - A local string variable for defining TTS parameters.
   - A local string variable for holding the reason that the playing of the synthesized speech was ended.
   - A local string variable for the log identifier associated with all log information created for the session.
   - A local string variable to hold the text string to be synthesized or a URL reference.
   - A local numeric variable to hold the Session ID.
   - A local numeric variable to hold the return code.

3. Establish a connection between the caller and WebSphere Voice Response for an incoming call:
   a. Select the AnswerCall icon in the PhoneLine folder of the Action Palette.
   b. Drag it to the work area.
   c. Configure the AnswerCall action. For more information on how to do this, refer to the “State table actions” chapter in the WebSphere Voice Response for AIX: Application Development using State Tables manual.

4. Create the required parameter lists and text strings for TTS configuration.
   a. Select the AssignData icon in the Variable folder of the Action Palette and drag it to the work area.
b. Click the Assign operator and select the PutTag parameter.

c. Click Const for Operand 1.

d. Type in the Operand 1 field an MRCP text-to-speech configuration parameter, for example the parameter speech-language.

e. Click Const for Operand 2.

f. Type in the Operand 2 field a valid value for the parameter in the Operand 1 field, for example, en-US.

g. Click Result to display the Variables window.

h. Click Local to display a list of previously defined local variables.

i. Click the name of the tagged string variable to use for specifying speech synthesis parameters for the session, and click Select to save the information and close the window. (The current parameter = value pair is appended to the tagged string variable.)

j. In the Action Assign Data window, type a meaningful description in the Description field to help you identify the Assign Data action.

k. Click OK save the information and close the window.

l. Repeat steps 4a on page 10 through 4k for each of the other parameter = value pairs defining TTS parameters that you want to include in the tagged string for the TTS parameters. See Table 1 on page 5 for details of possible speech synthesis parameters. You may need to adjust the values of these parameters to suit your synthesizer server.

5. Connect with the MST custom server by using Host -> OpenHostServerLink

   a. To enable your application to establish a link with the MST custom server, select the OpenHostServerLink icon in the Host folder of the Action Palette and drag it to the position (in the List View) of the AnswerCall action in the state table. (The OpenHostServerLink action is added below it.) Configure the OpenHostServerLink action.

6. Open a TTS session by using the MST_TTS_Assign state table.

   a. Double-click on the new InvokeStateTable action in the state table to display the Action InvokeStateTable window.

   b. Type MST_TTS_Assign in the State Label field.

   c. Type a meaningful description for the action in the Description field.

   d. Select the Invoke Direct radio button.

   e. Click on the State Table push button to display the State Table Selection window.

   f. Select MST_TTS_Assign from the list of available state tables displayed, and click on the OK push button to confirm your selection and close the window.

   g. Click on the Parameters push button to display the Invoke State Table Parameters window.
h. Select the **Const** radio button for **PARM1** and type in the field the name of the TTS engine to be used.

   **Note:** The name must match the type of an engine configured on an available TTS engine server for the TTS language and specified in the $CUR_DIR/ca/MST_dir/mst.xml configuration file.

i. Click on the **PARM2** push button to display the Variables window.

j. Select the **Local Variables** radio button.

k. Select the local variable for the TTS return code from the list of available local variables displayed, and click on the **Select** push button to confirm your selection and close the window. This variable is used to hold the numeric return code issued when the MST_TTS_Assign state table is invoked.

l. Click on the **PARM3** push button to display the Variables window.

m. Select the **Local Variables** radio button.

n. Select the local variable for the TTS session from the list of available local variables displayed, and click on the **Select** push button to confirm your selection and close the window. This variable is used to hold the numeric identifier of the MRCP speech synthesis session being started. It is assigned at runtime by the system.

o. Click on the **PARM4** push button to display the Variables window.

p. Select the **Local Variables** radio button.

q. Select the local variable for the TTS log identifier from the list of available local variables displayed, and click on the **Select** push button to confirm your selection and close the window. This variable is used to hold the string identifier that is associated with all log information that will be created for the MRCP speech synthesis session being started. It is assigned at runtime by the system.

r. In the Action InvokeStateTable window, leave the field for **Result 0** blank, but for each of the other possible **Result State Transfer** fields, type in the field the name of the label to use for the state table action that will handle such an error.

s. Click on the **OK** push button to save your changes and close the window.

7. You now need to deal with the return code issued when the MST_TTS_Assign state table is invoked. This involves switching to a state table action that ends the current MRCP speech synthesis session in the event of an error, or otherwise proceeding with the next action in the state table. To do this:

   a. Select the **Case** icon in the **Variables** folder of **Action Palette** and drag it to the position (in the List View) of the previous action in the state table.
b. Double-click on the Case action in the state table to display the Action Case window.

c. Type a meaningful description for the action in the Description field.

d. Click on the Input Variable push button to display the Variables window.

e. Select the Local Variables radio button.

f. Select the local variable for the TTS return code from the list of available local variables displayed, and click on the Select push button to confirm your selection and close the window. This logs the numeric return code issued when an MST_TTS_Assign state table is invoked.

g. For each possible return code (as documented in “MST_TTS_Assign” on page 19), type in the When field the numeric return code, and in the Goto field, the corresponding label of the action to which processing is redirected. Leave the Default field blank to ensure that if no errors are returned, processing continues with the next action in the state table.

h. Close the Action Case window by selecting the Default field and pressing the Enter (Return) key.

   Note: You can define the state table actions that handle the error processing later, but you must use the Goto labels you have specified.

8. Optionally, to override for this session the speech synthesis configuration parameters defined in the $CUR_DIR/ca/MST_dir/mst.xml configuration file, select the InvokeStateTable action, invoke the supplied state table MST_TTS_Config:

   a. Type MST_TTS_Config in the State Label field.

   b. Type a meaningful description for the action in the Description field.

   c. Select the Invoke Direct radio button.

   d. Click on the State Table push button to display the State Table Selection window.

   e. Select MST_TTS_Config from the list of available state tables displayed, and click on the OK push button to confirm your selection and close the window.

   f. From the Action InvokeStateTable window, click on the Parameters push button to display the Invoke State Table Parameters window.

   g. Click on the PARM1 push button to display the Variables window.

   h. Select the Local Variables radio button.

   i. Select the local variable for the TTS session ID from the list of available local variables displayed, and click Select to confirm your selection and close the window. This variable is used to hold the numeric identifier of the MRCP speech synthesis session being configured. It is assigned at runtime by the system.
Click on the PARM2 push button to display the Variables window.

Select the Local Variables radio button.

Select the local variable for the TTS parameter string from the list of available local variables displayed, and click Select to confirm your selection and close the window. This variable was defined earlier in step 2 on page 10 and is a tagged string including valid MRCP TTS configuration header fields to specify TTS session parameters.

Click on the PARM3 push button to display the Variables window.

Select the Local Variables radio button.

Select the local variable for the TTS return code from the list of available local variables displayed, and click on the Select push button to confirm your selection and close the window. This variable is used to hold the numeric return code issued when the MST_TTS_Config state table is invoked.

You now need to deal with the return code issued when the MST_TTS_Config state table is invoked. This involves switching to a state table action that ends the current MRCP speech synthesis session in the event of an error, or otherwise proceeding with the next action in the state table. See step 7 on page 12 for details of how to do this. The possible return codes are documented in "MST_TTS_Config" on page 20.

Synthesize speech by using the MST_TTS_Speak state table to 'speak' a text string or a text file. Using the InvokeStateTable action, invoke the supplied state table MST_TTS_Speak:

From the Action InvokeStateTable window, click on the Parameters push button to display the Invoke State Table Parameters window.

Click on the PARM1 push button to display the Variables window.

Select the Local Variables radio button.

Select the local variable for the TTS session ID from the list of available local variables displayed, and click Select to confirm your selection and close the window. This variable is used to hold the numeric identifier of the MRCP speech synthesis session being configured. It is assigned at runtime by the system.

Select the Const radio button for PARM2, and type ascii-string to specify the format of the text to be synthesized. The string is automatically enclosed in MRCP 'speak' tags before being sent to the MRCP TTS server by the application.

Click on the PARM3 push button to display the Variables window.

Select the Local Variables radio button.

Select the string that holds the prompt text to be synthesized.

Click on the PARM4 push button to display the Variables window.

Select the Local Variables radio button.
k. Select the local string variable for defining TTS parameters from the list of available local variables displayed, and click Select to confirm your selection and close the window. This variable was defined earlier in step 2 on page 10 and is a tagged string including valid MRCP TTS configuration header fields to specify TTS session parameters. If you wanted to override for this particular ‘speak’ action the parameters specified in the MST_TTS_Config state table you could do so by specifying a different tagged string.

l. Select the Const radio button for PARM5, and type NONE to specify that the playing of synthesized text cannot be interrupted. If you are using text-to-speech instead of a recorded prompt you could use barge-in to interrupt it by specifying the available option:

- ALL — to allow interruption by pressing any DTMF key

m. Click on the PARM6 push button to display the Variables window.

n. Select the Local Variables radio button.

o. Select the local variable for the TTS return code from the list of available local variables displayed, and click on the Select push button to confirm your selection and close the window. This variable is used to hold the numeric return code issued when the MST_TTS_Speak state table is invoked.

p. Click on the PARM7 push button to display the Variables window.

q. Select the Local Variables radio button.

r. Select the local variable for the TTS stop reason from the list of available local variables displayed, and click on the Select push button to confirm your selection and close the window. This variable is used to hold information returned from the server on why playing of synthesized speech was stopped.

11. You now need to deal with the return code issued when the MST_TTS_Speak state table is invoked. This involves switching to a state table action that ends the current MRCP speech synthesis session in the event of an error, or otherwise proceeding with the next action in the state table. See step 7 on page 12 for details of how to do this. The possible return codes are documented in “MST_TTS_Speak” on page 21.

12. Free the TTS session by using the MST_TTS_Free state table.

   a. Double-click on the InvokeStateTable action in the state table to display the Action InvokeStateTable window.

   b. Type MST_TTS_Free in the State Label field.

   c. Type a meaningful description for the action in the Description field.

   d. Select the Invoke Direct radio button.

   e. Click on the State Table push button to display the State Table Selection window.
f. Select \texttt{MST\_TTS\_Free} from the list of available state tables displayed, and click on the \texttt{OK} push button to confirm your selection and close the window.

g. Click on the \texttt{Parameters} push button to display the Invoke State Table Parameters window.

h. Specify local variables for each of the parameters, as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>PARM1</td>
<td>The local variable for the TTS session ID</td>
</tr>
<tr>
<td>PARM2</td>
<td>The local variable for the TTS return code</td>
</tr>
</tbody>
</table>

i. In the Action InvokeStateTable window, for each of the \texttt{Result State Transfer} fields, type in the field the name of the label to use for the state table action that will handle such an error.

j. Click on the \texttt{OK} push button to save your changes and close the window.

13. Log any error condition that may occur when the state table application is run. For each event that you want to log, select the \texttt{LogEvent} icon in the \texttt{Miscellaneous} folder of the \texttt{Action Palette} and drag it to the position (in the List View) of the \texttt{InvokeStateTable} action in the state table.

a. Double-click on the \texttt{LogEvent} action in the state table to display the Action LogEvent window.

b. Type a meaningful description in the \texttt{Description} field.

c. Select the \texttt{Const} radio button for \texttt{Private Event}, and type a meaningful string to identify the type of error. Leave the \texttt{System Event} as an undefined variable.

d. In each of the \texttt{Result State Transfer} fields, type the label of the state table action that will handle exit processing.

e. Click on the \texttt{OK} push button to save your changes and close the window.

14. Disconnect with the MST custom server by selecting the \texttt{CloseHostServerLink} icon in the \texttt{Host} folder of the \texttt{Action Palette} and dragging it to the position (in the List View) of the previous action in the state table. This closes an open link with the MST Custom Server.

a. Double-click on the \texttt{CloseHostServerLink} action in the state table to display the Action CloseHostServerLink window.

b. Type a label name for the action in the \texttt{State Label} field.

c. Type a meaningful description for the action in the \texttt{Description} field.

d. Click on the \texttt{Server Type} push button to display the Server Type Selection window.
e. Select **Custom Server** from the list displayed, and click on the **OK** push button to confirm your selection and close the window.

f. Click on the **Server Name** push button to display the Servers Selection window.

g. Select **MST** from the list displayed, and click on the **OK** push button to confirm your selection and close the window.

h. Click on the **OK** push button to save your changes and close the window.

15. To enable your application to exit from the state table, select the **ExitStateTable** icon in the **StateTable** folder of the **Action Palette** and drag it to the position (in the List View) of the previous action in the state table.

   a. Double-click on the **ExitStateTable** action in the state table to display the Action ExitStateTable window.

   b. Type a label name for the action in the **State Label** field.

   c. Select a value of 0 to return to the calling InvokeStateTable action.

   d. Click on the **OK** push button to save your changes and close the window.

16. Close the connection between the caller and WebSphere Voice Response.

17. Terminate the application. Optionally, to terminate your application completely, freeing any allocated resources, hanging up any open telephone lines, and dropping all host sessions, select the **CloseEverything** icon in the **Miscellaneous** folder of the **Action Palette** and drag it to the position (in the List View) of the previous action in the state table.

---

**Implementing the application**

It’s a good idea to keep all the components of the application together as application objects. Start by clicking **Applications → Applications** in the WebSphere Voice Response Welcome window. Then create a new application. Within the application, you can create all the voice segments, prompts, and state tables you need. You can include grammars and parameter files in the application by adding their AIX file specifications to it. For more information about using the Application windows, see the *WebSphere Voice Response for AIX: Designing and Managing State Table Applications* guide.

You can create a state table either by using the WebSphere Voice Response State Table window or by coding it in ASCII format using any editor you like, and then importing it into the State Table window, where it is converted into binary format. See the *WebSphere Voice Response for AIX: Application Development using State Tables* reference manual for details.
To implement the state table, you must have completed the prompts and voice segments. You should also ensure that you have a grammar.

Having created the state table and validated it, you can then start to test and debug the application. See Chapter 5, “Problem determination,” on page 25 for information on troubleshooting, and error messages.
Chapter 4. MRCP TTS state tables

This chapter lists the state tables required for using text to speech:

- “MST_TTS_Assign”
- “MST_TTS_Config” on page 20
- “MST_TTS_Speak” on page 21
- “MST_TTS_Free” on page 24

MST_TTS_Assign

Purpose

Use this state table to open a session with the MRCP text-to-speech server.

Parameters

Input

engine_type (String)

The type (name) of the TTS engine to be used. This should correspond to the type of an engine configured on an available TTS engine server. This state table only assigns a TTS engine. The mst.xml file will contain mapping between engine_type (for example, gbtts) and a specific MRCP server (for example, machine.hursley.ibm.com, port 554).

Output

return_code (Number)

Contains the result of the action. See below for details.

session_id (Number)

If successful a session identifier is returned. This is used on all subsequent actions to access the same session.

log_id (String)

If successful a log identifier is returned. This is used by the API as an identifier that is associated with all log information created for this session. It is provided for the application’s use to relate its logging for this session to that of the API.

Return codes

0 - Success

An engine of the specified type is now allocated.
1 - Maximum number of sessions already opened
The maximum number of sessions have already been opened

2 - No free engine
A free engine of the type specified could not be accessed on the
specified MRCP server

7 - Protocol error
The text-to-speech server has reported a protocol error. More details
about the exact error will be logged to the WebSphere Voice Response
error log.

9 - Parameter error
A parameter is in error (for example, a parameter is too long). More
details about the exact error will be logged to the WebSphere Voice
Response error log.

10 - Failed
An unexpected problem occurred. More details about the exact error
will be logged to the WebSphere Voice Response error log.

MST_TTS_Config

Purpose
Optionally use this state table to modify the configuration of a session with
the MRCP text-to-speech server. Configuration data applies to the session, not
to an engine. Configuration remains valid for the duration of the session.
Sequential invocation of this action within a session can be used to add to and
override configuration settings for the session.

Usage
MST_TTS_Config augments and updates the default configuration in file
$CUR_DIR/ca/MST_dir/mst.xml, which must contain a set of header fields to be
loaded into the TTS engine when first allocated.

Parameters

Input

session_id (Number)
The session identifier allocated on the successful call to
MST_TTS_Assign.

session_parameters (Tag String)
The MRCP TTS configuration updates to be applied to the session. To
build the string, use the PutTag operator of the AssignData state table
action. See [4i on page 11] details of how to use PutTag. The MRCP
header fields that can be specified in session_parameters are defined in
the MRCP specification and are passed unchanged to the MRCP
TTS server using the MRCP SET_PARAMS call. See Table 1 on page 5.

Output

return_code (Number)
Contains the result of the action. See below for details.

Return codes

0 - Success
The configuration parameters have been set successfully.

7 - Protocol error
The text-to-speech server has reported a protocol error. More details
about the exact error will be logged to the WebSphere Voice Response
error log.

8 - Invalid session
The session ID specified was not valid.

9 - Parameter error
A parameter is in error (for example, a parameter is too long). More
details about the exact error will be logged to the WebSphere Voice
Response error log.

10 - Failed
An unexpected problem occurred. More details about the exact error
will be logged to the WebSphere Voice Response error log.

4nn Protocol error returned by the MRCP server if a parameter passed in
the correct format, but the MRCP server cannot process it, for
example, when an unsupported language is specified. For details of
these codes and their meaning, refer to the MRCP V1 specification
available at:


MST_TTS_Speak

Purpose

Use this call to synthesize and play a text string.

Parameters

Input
**session_id (Number)**

The session identifier allocated on the successful call to MRCP_TTS_Assign.

**text_type (string)**

This indicates the format of the text to be played, the possible values being:

- **ascii-string**
  
  The text is contained in the 'text' input parameter and will automatically be enclosed in MRCP <speak> tags by WebSphere Voice Response.

- **ascii-file**
  
  The text is contained in a file, the name (path) of which is specified in the 'text' input parameter. A file can only be loaded from a local storage device, and not an HTTP server.

- **xml-string**
  
  The string in 'text' is assumed to be formatted 'speak' XML which is sent to the TTS engine without any retagging

- **xml-file**
  
  The string in 'text' is assumed to be the name (path) of a file containing pre-formatted 'speak' XML

**text (String)**

See the description of 'text_type' above.

Note that the text string has a limit of 2000 characters. To play longer text strings, your application needs to break them up into smaller chunks or use one of the file modes of operation.

**parameters (Tagged String)**

The MRCP configuration to be used for this speak action only. These header fields will augment and update any session values set using MRCP_TTS_CONFIG or the configuration in file $CUR_DIR/ca/MST_dir/mst.xml for this action only. To build the string, use the PutTag operator of the AssignData state table action. See [4i on page 11](#) for details of how to use PutTag. The header fields that can be specified in this field are defined in the MRCP specification and are passed unchanged to the MRCP TTS server on the MRCP SPEAK call. Examples are as follows:

- Speech-Language en-us
- Voice-gender female
- Prosody-volume 25

**stop_keys (String)**

This attribute controls how the playing of synthesized text can be interrupted by the pressing of DTMF keys. The possible values are:
NONE
This is the default value and indicates that the playing cannot be interrupted. It is equivalent to FORCE_PLAY for the PlayVoiceFromHost action.

ALL Any DTMF key can interrupt the play. It is equivalent to STOP_PLAY_ON_DTMF for the PlayVoiceFromHost action.

Output
return_code (Number)
Contains the result of the action. See below for details.

stop_reason (String)
If the return_code is 0, this indicates what if anything caused the TTS play to stop. If it completed without interruption then this string is empty, otherwise it is one of the following:

DTMF: DTMF was detected.

HUP: Caller has hung up.

Return codes
0 - Succeeded
The TTS synthesized and played all or some of the prompt without any problems. Check the stop_reason to see what if anything caused the prompt to be stopped prematurely.

7 - Protocol error
The text-to-speech server has reported a protocol error. More details about the exact error will be logged to the WebSphere Voice Response error log.

8 - Invalid session
The session ID specified was not valid.

9 - Parameter error
A parameter is in error (for example, a text string exceeds 2000 characters in length). More details about the exact error will be logged to the WebSphere Voice Response error log.

10 - Failed
An unexpected problem occurred.

4nn Protocol error returned by the MRCP server if a parameter passed in the correct format, but the MRCP server cannot process it, for example, when an unsupported language is specified. For details of these codes and their meaning, refer to the MRCP V1 specification available at:
MST_TTS_Free

Purpose

Use this state table to close a session with an MRCP Server TTS engine. The resource should be explicitly freed when no longer required.

All resources must be freed before the application terminates otherwise the MST custom server will raise an error and may need to be stopped and restarted.

Parameters

Input

`session_id` (Number)
- The session identifier allocated on the successful call to `MST_TTS_Assign`.

Output

`return_code` (Number)
- Contains the result of the action. See below for details.

Return codes

0 - Succeeded Session closed.
- The TTS engine has been freed and may be allocated to another application.

8 - Invalid session
- The session ID specified was not valid.

10 - Failed
- An unexpected problem occurred. More details about the exact error will be logged to the WebSphere Voice Response error log.
Chapter 5. Problem determination

Return codes

0 - Success
   The operation succeeded.

1 - Maximum number of sessions already opened
   The maximum number of sessions have already been opened.

2 - No free engine
   A free engine of the type specified could not be accessed on the
   specified MRCP server.

7 - Protocol error
   The text-to-speech server has reported a protocol error. More details
   about the exact error will be logged to the WebSphere Voice Response
   error log.

8 - Invalid session
   The session ID specified was not valid.

9 - Parameter error
   A parameter is in error (for example, a parameter is too long or not
   set). More details about the exact error will be logged to the
   WebSphere Voice Response error log.

10 - Failed
    An unexpected problem occurred. More details about the exact error
    will be logged to the WebSphere Voice Response error log.

11 - No Result Available
    No result is available.

12 - Invalid State
    The application is not in the correct state to issue this API.

13 - Bad Complete Message
    An MRCP Complete message contained a bad value.

15 - Message Timed Out
    A response message from the MRCP Server was not received in a
    timely manner. The associated session has been closed.

101 - State Table Assign Failed
    A state table assign action failed.
102 - MRCP_GetPort Error
A problem occurred within the MRCP_GetPort state table. More
details about the exact error will be logged to the WebSphere Voice
Response error log.

104 - Host Not Open
A required custom server has not been opened or is not running.

105 - Timeout
A timeout occurred on a state table ReceiveData action.

106 - ReceiveData Error
An error occurred on a state table ReceiveData action.

400-499 - MRCP Message Completion Codes
MRCP Message completion codes are passed through for application
handling. For details of these codes and their meaning, refer to the
MRCP V1 specification available at:

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For country-specific notes on the use of WebSphere Voice Response, refer to the README file located in the directory /usr/lpp/dirTalk/homologation. The file name is in the format README_homologation.xxxx, where xxxx is the country/region identifier.

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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.

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.

C

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.
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](#).

CGI  See [Common Gateway Interface](#).

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](#), [music channel](#).

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](#) (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).

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](#).

CIC  See [circuit identification code](#).

CICS  See [customer information control system](#).

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 file
See parameter file.

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.

container window
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-canceled voice data that can then be used by the engine in a recognition attempt. This is similar to barge-in.

D

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 Kb 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 being 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 callers 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 PBXs 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

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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` and `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. (Compand 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.

network 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+124B).

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.

pushbutton telephone
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.

**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.

**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.
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.

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.

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.
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**, and **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 E1.

**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 Kbps 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 Kbps 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.

**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](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

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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