

# IBM WebSphere Voice Server for Multiplatforms V5.1.1/V5.1.2 and WebSphere Voice Response V4.2

## An Interoperability Guide

Use WebSphere Voice Server  
Connector for interoperability

Apply best practices to  
VoiceXML development

Test and debug  
VoiceXML applications



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**IBM WebSphere Voice Server for Multiplatforms  
V5.1.1/V5.1.2 and WebSphere Voice Response V4.2:  
An Interoperability Guide**

March 2006

**Note:** Before using this information and the product it supports, read the information in “Notices” on page v.

**First Edition (March 2006)**

This edition applies to Version V5.1.1 and V5.1.2 of WebSphere Voice Server and V4.2 of WebSphere Voice Response for AIX.

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# Preface

Next generation IBM® WebSphere® Voice Server V5.1.1 and V5.1.2 products leverage the power of IBM WebSphere Application Server, the IBM flagship product and leader in the industry. WebSphere Voice Server V5.1 is the first release of a novel (patent pending) architecture for speech servers, based on WebSphere Application Server. No other speech server in the industry can match the on demand capabilities that WebSphere Voice Server can provide as a result of its WebSphere Application Server base. Add the power of the Java™ 2 Enterprise Edition (J2EE™) platform, including the runtime, services, and container management, and the result is a powerful speech server that integrates into existing IT infrastructures. Having voice on the same network as visual and data applications greatly reduces the initial speech investment. It simplifies management and achieves improved integration with customer content, which is a feat that no other speech vendor can match.

WebSphere Voice Response for AIX® V4.2, with the WebSphere Voice Server Connector software, delivers next generation voice processing that meets advanced call-routing and voice self-service needs for a contact center enterprise. It uses standard IBM Quad Digital Trunk Telephony Adapter (DTTA) or Digital Trunk Extended Adapter (DTEA) hardware for telephony support and a VoiceXtensible Markup Language (VoiceXML) V2.1 compliant browser for application programming interfaces.

IBM WebSphere Voice Server V5.1.1 and V5.1.2 and the WebSphere Voice Response for AIX V4.2 products support full open standards using:

- ▶ Speech Recognition Grammar Specification (SRGS) V1.0 for grammars
- ▶ Semantic Interpretation for Speech Recognition (SISR) for semantic interpretation
- ▶ Speech Synthesis Markup Language (SSML) V1.0 for Text to Speech markup
- ▶ VoiceXML V2.0 and V2.1 for speech application markup
- ▶ Real-Time Streaming Protocol (RTSP) and Real-Time Protocol (RTP) for interoperability between the interactive voice response (IVR) and speech server

This IBM Redpaper is a guide for integrating IBM WebSphere Voice Server V5.1.1 and V5.1.2 and WebSphere Voice Response for AIX V4.2. It explains how you can tailor and configure WebSphere Voice Server and WebSphere Voice Response in simple and complex topologies. It presents the best practices as applied to the Voice User Interface (VUI) design. This paper guides you through the steps to implement and deploy a simple voice-enabled application using WebSphere Voice Response for AIX V4.2 and WebSphere Voice Server V5.1.1 and V5.1.2.

In addition, this paper explains how you can develop, test, and deploy a simple voice-enabled application using VoiceXML. And it demonstrates how you can use automatic speech recognition (ASR) and text-to-speech (TTS) voice technologies.

Prior to reading this paper, you must have a basic knowledge of voice-enablement of applications using VoiceXML, WebSphere Voice Response, WebSphere Voice Server, and WebSphere Application Server.

## The team that wrote this Redpaper

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# Introduction to IBM WebSphere Voice products

This chapter introduces you to the IBM WebSphere Voice Server for Multiplatforms V5.1.1 and V5.1.2 products with the IBM WebSphere Voice Response for AIX V4.2 product. These products interoperate using the WebSphere Voice Server Connector. Both products strongly support open standards such as VoiceXtensible Markup Language (VoiceXML) V2.0 and V2.1, Speech Recognition Grammar Specification (SRGS) V1.0, Semantic Interpretation Speech Recognition (SISR) V1.0, and Speech Synthesis Markup Language (SSML) V1.0. Open standards allow customers to reach new levels of innovation while continuing to deliver the increased productivity that is necessary to improve the bottom line.

## 1.1 Overview of WebSphere Voice Server for Multiplatforms

IBM WebSphere Voice Server V5.1.1 and V5.1.2 leverage the power of WebSphere Application Server. WebSphere Application Server is the IBM flagship product and leader in the industry. WebSphere Voice Server V5.1 is the first release of a novel (patent pending) architecture for speech servers, based on WebSphere Application Server. Combining IBM WebSphere Voice Server and WebSphere Application Server means that speech customers can take advantage of the on demand features of WebSphere Application Server. Using these features is critical to the nature of self-service applications with speech.

The features of WebSphere Voice Server include:

- ▶ Load balancing
- ▶ Scalability
- ▶ Failover
- ▶ Recovery
- ▶ Maintenance
- ▶ Serviceability
- ▶ Management
- ▶ Administration

No other speech server in the industry can match the on demand capabilities that WebSphere Voice Server is able to provide as a result of its WebSphere Application Server base. WebSphere Voice Server V5.1.1 and V5.1.2 run as a Java 2 Enterprise Edition (J2EE) Application inside WebSphere Application Server. The result is a powerful speech server that integrates into existing IT infrastructures. Having voice on the same network as visual and data applications greatly reduces the initial speech investment. It simplifies management and achieves improved integration with customer content, which is a feat that no other speech vendor can match.

### 1.1.1 WebSphere Voice Server V5.1.1 and V5.1.2 features

The following sections briefly highlight the WebSphere Voice Server features. Refer to *IBM WebSphere Voice Server for Multiplatforms V5.1.1/V5.1.2 Handbook*, SG24-6447, for a comprehensive description with examples and instructions for usage.

#### **Easier to use administration**

Making today's speech applications easier to use is critical. Speech solutions requiring minimal time and skill to start and run (installation, configuration) the speech system will save development dollars and allow for rapid return-on-investment (ROI). When problems arise, system administrators are expected to quickly identify and resolve system problems. Downtime must be minimized if a system or application problem occurs.

To administer, configure, and monitor WebSphere Voice Server, you can use the following tools:

- ▶ Administrative console
- ▶ Command line tools
- ▶ Scripts

The Administrative Console application is a Web application that runs on a WebSphere Application Server for administration and monitoring of resources. On a WebSphere Voice Server, an additional view is added to the administrative console to allow monitoring of speech resources. System administrators can obtain "at-a-glance" knowledge of their system by reviewing a single interface. It's also a centralized configuration tool.

The console provides the following features:

- ▶ Views of overall system status and detailed status of automatic speech recognition (ASR) and text-to-speech (TTS; a technology that synthesizes spoken language from text) resources
- ▶ Views of general resources such as system status and free memory
- ▶ Capability for administrators to start and stop voice servers
- ▶ Capabilities for administrators to configure the ASR and TTS engines, preloaded grammars, lexicons, and installed voices

Command line tools are provided to start and stop servers, add or remove servers to a cell, and obtain the status of a server.

The WebSphere Application Server `wsadmin` tool provides the ability to run scripts. You can use the `wsadmin` tool to manage a WebSphere Voice Server installation, configuration, application deployment, and server runtime operations. The WebSphere Application Server supports Java Command Language (JACL), an alternate implementation of the interpreted script language, Tool Command Language (TCL), written in Java.

### **Open standards supported**

Standards provide flexibility in creating speech architectures for customers. Standards allow greater reuse of a customer's existing infrastructure, thereby lowering the costs of speech projects. The following standards are supported:

- ▶ Media Resource Control Protocol (MRCP) Version 1 draft 4
- ▶ VoiceXML V2.0 and V2.1 specification
- ▶ VoiceXML V2.0 Built-in Grammar Types for all languages
- ▶ Speech Recognition Grammar Specification V1.0  
<http://www.w3.org/TR/2004/REC-speech-grammar-20040316>
- ▶ Speech Synthesis Markup Language V1.0  
<http://www.w3.org/TR/2004/REC-speech-synthesis-20040907>
- ▶ Semantic Interpretation for Speech Recognition - W3C Working Draft 1, dated April 2003  
<http://www.w3.org/TR/2003/WD-semantic-interpretation-20030401>

### **Supported operating systems**

The following operating systems are supported:

- ▶ Microsoft® Windows® Server 2003
- ▶ Red Hat Enterprise Linux WS/ES/AS V3.0 Update 1 for Intel® platforms
- ▶ SUSE Linux Enterprise Server (SLES) V8.0 with SP2a or SP3

For a list of the current platforms supported, see the WebSphere Voice Server Web site at:

[http://www.ibm.com/software/pervasive/voice\\_server/system\\_requirements](http://www.ibm.com/software/pervasive/voice_server/system_requirements)

## Supported languages

WebSphere Voice Server V5.1.x supports several different languages. All languages have been extensively tested with WebSphere Voice Response. Table 1-1 lists the supported language ASR and TTS and Concatenative Text to Speech (CTTS) available. WebSphere Voice Server uses the third generation of CTTS.

Table 1-1 Supported languages for WebSphere Voice Server V5.1.x

Language	WebSphere Voice Server V5.1.1 (Linux) and V5.1.2 (Microsoft Windows Server 2003) ASR support	WebSphere Voice Server V5.1.1 (Linux) and V5.1.2 (Microsoft Windows Server 2003) CTTS support	
		Female	Male
United States (US) English	Yes	Yes	Yes
United Kingdom (UK) English	Yes	Yes	Yes
Australian (AUS) English	Yes	Yes *Uses UK English CTTS	Yes *Uses UK English CTTS
Canadian French	Yes	Yes	Yes
German	Yes	Yes	Yes
Latin American Spanish	Yes	Yes	Yes
Japanese	Yes	Yes	Yes
Simplified Chinese	Yes	Yes	No

WebSphere Voice Server V5.1.1 and V5.1.2 have the restriction of one language per WebSphere Voice Server machine.

**Attention:** WebSphere Voice Server V5.1.3 lifts the restriction of one language per WebSphere Voice Server machine and now supports multiple languages per machine.

Check the IBM WebSphere Voice Server Web site (listed in the previous section) for the list of current languages supported.

### Text to speech

WebSphere Voice Server fully supports the SSML standard. TTS supports lexicons for specifying pronunciations. Large lexicon files may be preloaded at WebSphere Voice Server startup and persisted for all phone calls.

A voice can be specified by its voice-name attribute or by gender. Or it can default to the specified voice on the WebSphere Voice Server machine per language. For each voice, the application developer can control the various prosody values.

TTS also supports embedded audio files within the SSML text.

### Automatic speech recognition

WebSphere Voice Server V5.1.1 and V5.1.2 provide robust speech recognition capabilities with silence and speech detection support that reduce false barge-ins in noisy environments. ASR supports the use of lexicons to provide additional high quality pronunciations.

## 1.1.2 Overview of WebSphere Application Server Network Deployment

WebSphere Voice Server provides customers with a portion of IBM WebSphere Application Server Network Deployment V5.1.1. This portion is provided as part of the WebSphere Voice Server base license with no additional charge.

WebSphere Voice Server includes the following features of WebSphere Application Server Network Deployment:

- ▶ WebSphere Application Server with IBM HTTP Server
- ▶ WebSphere Edge Server Load Balancer component
- ▶ WebSphere Application Server Deployment Manager

## 1.1.3 WebSphere Application Server with IBM HTTP Server

WebSphere Voice Server uses WebSphere Application Server as its infrastructure, running as a J2EE Enterprise Application within WebSphere. WebSphere Voice Server provides additional Administrator Console extensions to allow “at-a-glance” system monitoring. Additional panels allow you to change configurations. In addition, you can diagnose and troubleshoot problems. You can start and stop WebSphere Voice Server machines from the Administrator Console or via WebSphere Application Server’s administrative scripting via JACL. This feature allows the automation of administrative tasks.

WebSphere Application Server allows system administrators to use the same administration capabilities for the user interface or speech components of Tier 1, the customer application server (such as VoiceXML) of Tier 2, and the customer content server (such as the database) of Tier 3.

The following features are provided:

- ▶ Common Administrative Console
- ▶ Runtime error, warning, and informational alarms
- ▶ Ability to enable trace and troubleshoot in real-time
- ▶ Detailed Runtime panel for seeing CPU and memory, status of ASR and TTS engines, and their current allocation
- ▶ Detailed Configuration panel to easily change the most important WebSphere Voice Server parameters
- ▶ Ability to stop and restart WebSphere Voice Server machines as needed

## 1.1.4 WebSphere Edge Server Load Balancer component

WebSphere Voice Server uses Load Balancer as the single point of access to your WebSphere Voice Server farm from the interactive voice response (IVR). Load Balancer provides on demand speech technology. On demand is not about technology for the sake of technology; it’s about enabling new ways of doing business so that your company has the ability to respond to any customer demand, market opportunity, or external threat.

In today’s competitive business, a lack of system availability means a loss of customer satisfaction impacting your revenue. Many companies operate 24x7 so the days of scheduled downtime, where the application is not available, are no longer acceptable. If a critical application is not available, end users will become frustrated and “shop” around. Failover and recovery are no longer luxuries but requirements.

The Load Balancer provides capabilities for continuous access to speech not only encompassing fail-over and recovery, but providing full serviceability to take a machine

gracefully out of service with no impact to availability or loss of speech requests. The Load Balancer allows for adding or removing WebSphere Voice Server machines on demand without changing the configuration on WebSphere Voice Response or on any other machine except for the Load Balancer.

This feature provides horizontal scalability. The Load Balancer allows WebSphere Voice Server machines on differing operating systems or different hardware capabilities to be assigned a “weight” based upon its capabilities. For example, a quad processor machine can be assigned a “weight” where it receives twice as much work as a dual processor machine.

The Load Balancer can be seamlessly added to your existing network to benefit from all of its features. Its easy to install and configure, and it uses standard TCP/IP protocols. In addition, you can configure the Load Balancer for full high availability.

The Load Balancer software is lightweight and can run on a low-end machine. It has a number of useful features:

- ▶ It detects whether a voice server is unavailable or inaccessible and does not continue to send messages to this server.
- ▶ It uses the number of new, active, and complete connections to determine which server should receive the message.
- ▶ The addition of a ping advisor allows the Load Balancer to more accurately determine when a server fails or recovers.
- ▶ If one voice server fails, or needs to be taken out of service, the other servers dynamically take over the load.
- ▶ Voice servers can be dynamically added to the Load Balancer, providing the scalability to increase the number of engines available. You only have to change the Load Balancer configuration. No changes are required on WebSphere Voice Response or any other Voice server machine.
- ▶ It assigns a “weight” to each voice server based upon its hardware capabilities. This allows a cluster to be made up of a combination of high and low end machines. For example, a quad processor may be assigned a weight double that of a dual processor machine.
- ▶ It gracefully takes a voice server out of service for maintenance without losing any calls, and the work on that voice server completes normally. Because the “weight” is zero, the Load Balancer no longer sends the machine any new work. When the active number of connections is zero, the machine can be taken out of service.

### 1.1.5 WebSphere Deployment Manager

WebSphere Voice Server uses Deployment Manager to create a network of WebSphere Voice Server systems that are configured to run as a group known as a *cell*. The Deployment Manager capability provides centralized configuration, administration, and workload management for the entire cell. One computer is designated as the central Deployment Manager machine to oversee the cell from a single Administrative Console. This Administrative Console is an extension of the classic WebSphere Application Server console.

Each Deployment Manager manages one cell. When a voice server is added to this cell, it is referred to as a *node*. Each Deployment Manager can manage multiple nodes, which can be installed on different operating systems.

Deployment Manager can run on a low-end machine because it is lightweight. You can install Deployment Manager on the same machine with Load Balancer.

The use of Deployment Manager in a voice server deployment provides several benefits to the administrator:

- ▶ Multiple WebSphere Voice Server machines can be managed from a single location, either as a group or independently.
- ▶ Servers can be brought in or out of operation using the Administrative Console on the Deployment Manager.
- ▶ You can see at a glance which servers are running and which are stopped.
- ▶ Support for JACL scripts enables you to automate common system administration functions.
- ▶ Via WebSphere Voice Server console extensions, system administrators can see the CPU, memory and active status of ASR and TTS engines on any WebSphere Voice Server machine in the cell.

### 1.1.6 Tools and components

WebSphere Voice Server includes the Voice Toolkit for WebSphere Studio. The Voice Toolkit for WebSphere Studio V5.1 is based on Eclipse and integrates into WebSphere Studio Application Developer or WebSphere Site Studio Developer V5. The Voice Toolkit V6 builds upon IBM Rational® Application Developer or Rational Web Developer and is available as a free download. Both sets of products provide a single tooling strategy for voice and data applications, speeding the development and testing of grammars, lexicons, troubleshooting, and analysis. Customers can deploy their applications faster, using common tooling with Eclipse. Application developers don't have to learn a new tool for each type of application.

In particular, the Voice Toolkit for WebSphere Studio provides graphical user interface (GUI) tools for:

- ▶ Editors to test grammar using speech, text, or enumeration with the ability to listen to each synthesized request and adjust the results
- ▶ A Grammar Development Editor that allows rapid SRGS XML or ABNF grammar development
- ▶ A Lexicon Development Editor for building pronunciations
- ▶ A Grammar Test Tool that allows for batch testing of grammars for incremental improvement
- ▶ A Voice Server Trace Analysis tool for troubleshooting, analysis, and tuning of your speech applications available in the Voice Toolkit V6

## 1.2 Overview of WebSphere Voice Response for AIX V4.2

WebSphere Voice Response for AIX V4.2 with the WebSphere Voice Server Connector can support multiple languages. This support allows for the ability to have one Load Balancer serving multiple clusters, where each cluster uses a different language. WebSphere Voice Response can be configured to use different cluster addresses for each language.

### 1.2.1 Redpaper lab environment

The lab environment used for this Redpaper is illustrated in Figure 4-1 on page 54. The Gordon Kapes System 930 Telephony Simulator replaces the PBX and is configured as a DMS 100 ISDN-PRI to emulate a Central Office switch.

IBM WebSphere Voice Response V4.2 was installed on a AIX V5.2 IBM @server pSeries® server with an IBM Arctic Digital Trunk X Adapter (DTXA) telephony board with a four trunk channelized ISDN PRI/T1 interface. The WebSphere Voice Server Connector was installed to allow interoperability with WebSphere Voice Server.

## 1.2.2 Typical interactive voice response scenario

Typically, business applications that are enabled for voice are either inquiry or transaction-based functions or applications.

For example, a customer can invoke a query application to retrieve information from a database. Account balances, weather reports, movie listings, or stock quotes can now be obtained from a customer contact center's IVR. These are common examples of query applications that can be enabled for voice. Usually, the application guides the customer through a series of menus and forms aided by instructions, prompts, and choices. The application uses either prerecorded audio files or TTS. The customer can use spoken words to make selections and provide information. When the database records are retrieved, the system presents the information by playing back audio files.

A transaction application is used for the purpose of performing specific tasks. Transferring funds between accounts and the buying and selling of stocks are popular examples of this type of application. In the previous scenarios, the application starts by authenticating the customer through an identification process. Then it uses a series of windows to guide the customer to enter the required data such as the source and target accounts for a funds transfer transaction. The system plays instructions, prompts, and available options using prerecorded audio files or by performing text to speech. The customer responds using spoken commands. When the required data is collected, the system performs the transaction and returns the result.

This functionality is achieved through the use of VoiceXML applications. With WebSphere Voice Response, VoiceXML applications can be developed, tested, and deployed using IBM WebSphere Voice Toolkit. In addition, the WebSphere Voice Toolkit has critical grammar and lexicon development utilities, as well as a grammar test tool for rapid grammar development.



## Architecture and deployment concepts

An interactive voice system is typically comprised of a few key elements whose objective is to receive, respond to, and route call and data traffic. Three important elements to consider are the interactive voice response, Voice Server and Application Server. Each element needs its own machine to ensure proper scalability and failover and recovery requirements.

WebSphere Voice Response V4.2 acts as the interactive voice response element running on AIX V5.2 on pSeries servers. WebSphere Voice Server V5.1.2 provides the speech services, namely automatic-speech-recognition (ASR) and text-to-speech (TTS) capabilities. It runs on Red Hat Linux Enterprise server V3 or SUSE Linux Enterprise Server V8 and Microsoft Windows Server 2003 Enterprise or Standard Editions. And WebSphere Application Server runs on a variety of hardware and software platforms.

## 2.1 Deployment objectives

WebSphere Voice Response is the first component to respond to the user. It essentially answers the phone and provides the caller with choices that typically can be either spoken or selected using the telephone keypad. WebSphere Voice Response acts as the voice driven browser for retrieval of control and content documents that might reside on a separate application server. The WebSphere Voice Response voice browser then reads VoiceXtensible Markup Language (VoiceXML) applications that reside on an application server fetched via HTTP requests. The application server could be located anywhere, local or remote. However, considering the type and weight of data to be transferred, it may be necessary to keep the application server and voice browser within the same controlled network.

WebSphere Voice Response uses the WebSphere Voice Server Connector to communicate with WebSphere Voice Server. This is only possible for applications written in VoiceXML. State Table and Java applications may not access the WebSphere Voice Server V5.1.x capabilities.

### 2.1.1 Voice applications

The effectiveness of a voice driven environment, such as the one described in the previous section, depends on the development of voice applications that consist of VoiceXML documents and the content required by them, such as grammars, lexicons, and audio prompts. From that perspective, voice applications are essentially similar to visual Web content, except that the browser is a server-side application.

The primary system behavior is defined in the voice applications. The WebSphere Voice Response and WebSphere Voice Server components merely support the applications by providing services that enable the communication with the end user. The application's role is to extend its capabilities such as communicating with databases or other sources.

The ensemble of all three WebSphere products supporting the Voice Application grants both vertical and horizontal scalability that is not found in other products. If used together, the WebSphere Voice Response, WebSphere Voice Server, and WebSphere Application Server provide a foundation for applications and voice services to be launched, with each component having their own qualities and extensions.

### 2.1.2 Application flow

A voice application follows this sequence of actions to complete a simple transaction:

1. A call is received by the WebSphere Voice Response.
2. The voice browser consults the configuration file, `default.cff`, that points the call to either a VoiceXML or Call Control Extensible Markup Language (CCXML) application that resides on the WebSphere Application Server.
3. The VoiceXML browser executes the initial prompt in the VoiceXML application. The VoiceXML application may request that an audio file be fetched from the WebSphere Application Server and then played to the user. The VoiceXML application can also request text be spoken to the user by making a TTS request to the WebSphere Voice Server. The WebSphere Voice Response configuration file, `default.cff`, specifies the WebSphere Voice Server resource details based on the required language and type of resource, that is TTS.
4. Depending on the application, the user is prompted to make a choice. At that time, the voice browser again consults the WebSphere Voice Response configuration file,

default.cff, that redirects the incoming user utterance to the ASR engine in the WebSphere Voice Server.

5. Having interpreted the audio, the WebSphere Voice Server returns the recognition result to WebSphere Voice Response where the VoiceXML browser references the VoiceXML application for the appropriate action to take. Example actions are that the field is filled (a match in the grammar), a no-match has occurred (spoken utterance did not match the grammar), or a no-input occurred (the user did not respond to the prompt).

Now one transaction between the user and voice application is complete.

During a transaction, a TTS engine may access a lexicon to produce better pronunciation. An ASR engine requires a grammar or grammars. Optionally the grammar may reference a lexicon or lexicons to interpret the correct user utterances. By preloading large grammars and lexicons, WebSphere Voice Server responds faster as illustrated in Figure 2-1.

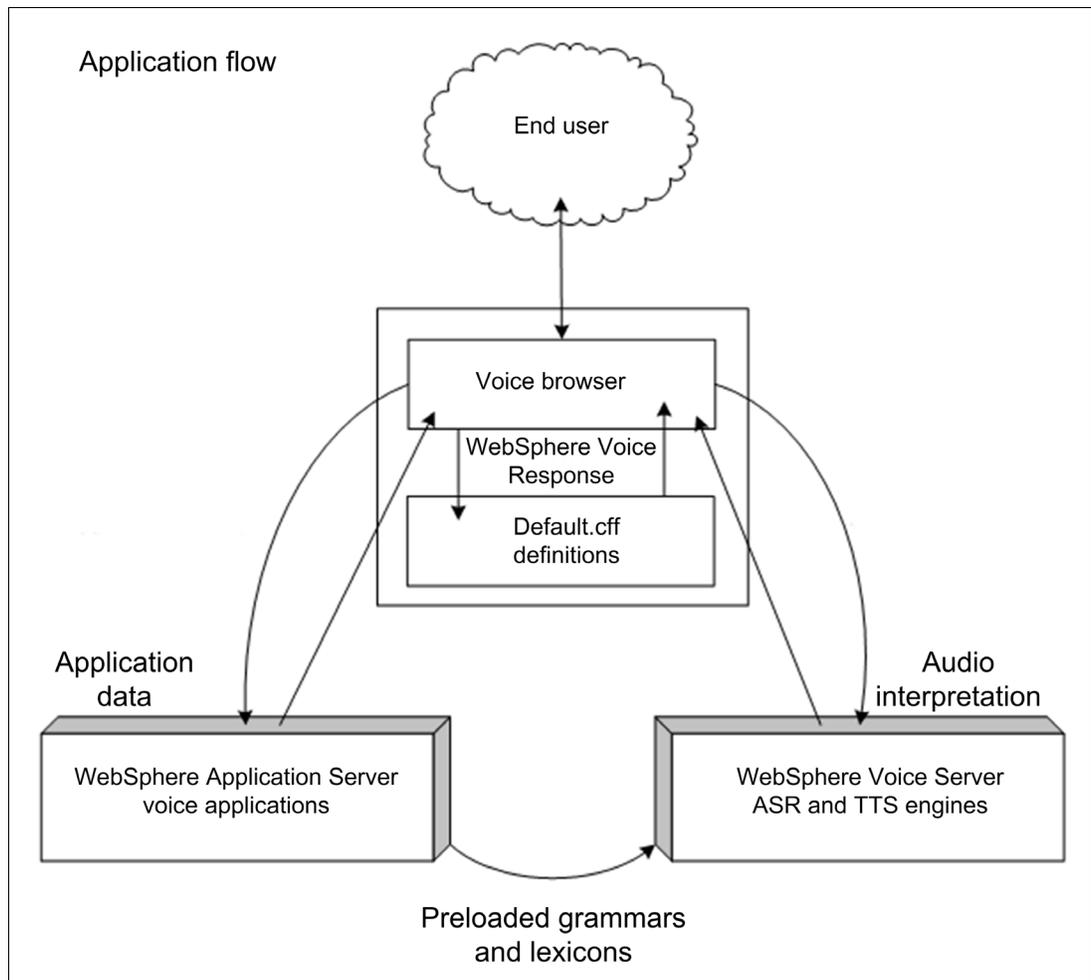


Figure 2-1 Preloaded grammars and lexicons

## 2.2 Basic deployment solution

A basic deployment solution with scalability considerations should consist of two machines:

- ▶ A pSeries machine running WebSphere Voice Response and WebSphere Application Server
- ▶ An Intel machine running WebSphere Voice Server

Also consider accessibility to the Voice Toolkit for WebSphere Studio for the development of suitable voice applications for this environment.

For a basic deployment solution and to keep things simple, as illustrated in Figure 2-2, we chose to store our VoiceXML applications on the WebSphere Voice Response server. Static VoiceXML files can be served over HTTP using the IBM HTTP Server or a separately licensed WebSphere Application Server.

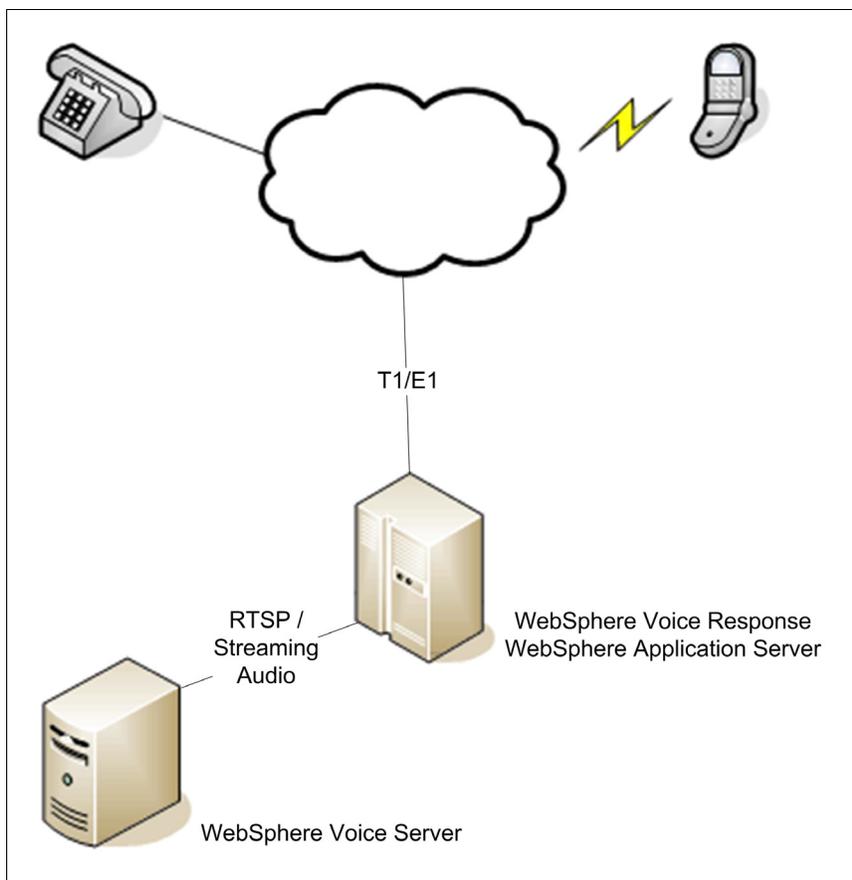


Figure 2-2 Basic deployment solution topology

## 2.3 Advanced deployment solution

An advanced infrastructure capable of supporting a high call volume can consist of multiple WebSphere Voice Response and WebSphere Voice Server machines. The demands of the application determine the WebSphere Application Server requirements.

### 2.3.1 WebSphere Edge Server Load Balancer component

For our purposes, it is possible to demonstrate the high volume capacity by increasing the number of WebSphere Voice Server machines. We can manage them as a cluster by using the WebSphere Edge Server Load Balancer component shipped with the WebSphere Voice Server package on the Deployment Manager CD.

In addition to using Deployment Manager to manage all Voice Servers from a single Administrative Console, the Load Balancer component can provide scalability, maintainability, failover, and recovery capabilities between the WebSphere Voice Server and WebSphere Voice Response systems. If call volume increases, an additional WebSphere Voice Server may be added to the Load Balancer configuration and brought online to start receiving ASR or TTS requests.

### 2.3.2 Deployment Manager

The Deployment Manager provides a single Administrative Console to manage all WebSphere Voice Server machines. It requires a light-weight machine and can be installed on any machine in the network.

### 2.3.3 Single system image

As shown in Figure 2-3, the WebSphere Voice Response server communicates with the WebSphere Voice Server cluster through the Load Balancer component. Meanwhile the WebSphere Application Server plays a dual role of providing content as well as running the Deployment Manager. The Deployment Manager needs to run only while configuring the cell. The Load Balancer component provides a single interface for WebSphere Voice Response to the WebSphere Voice Server machines.

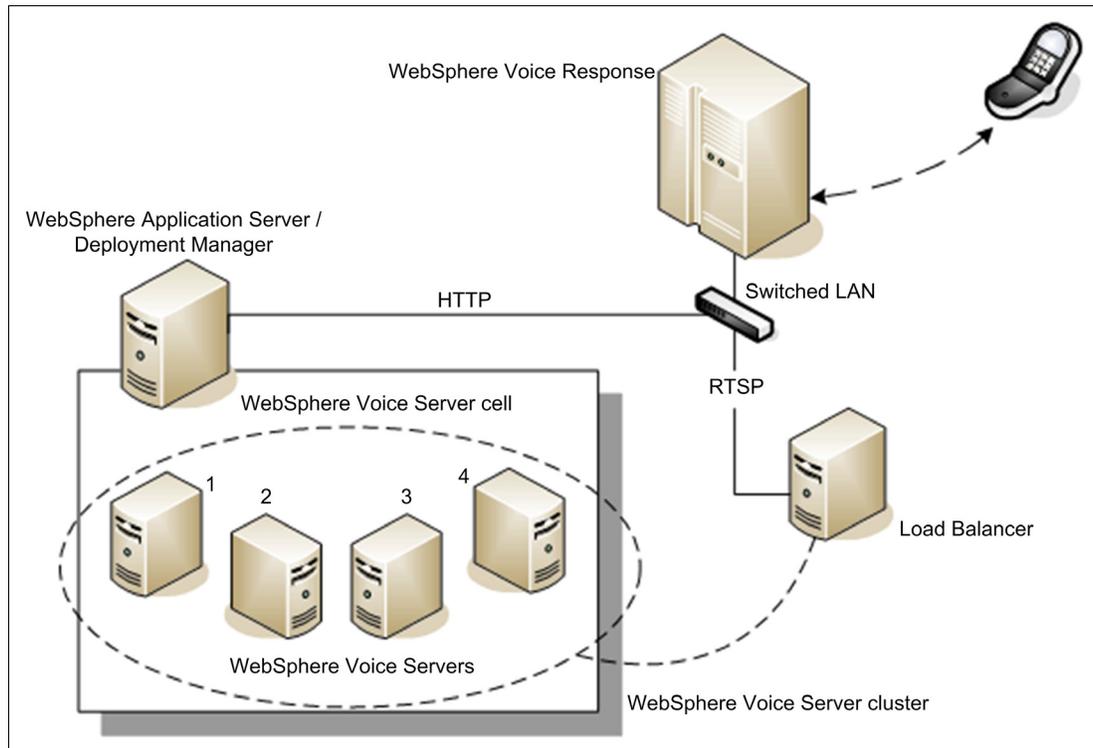


Figure 2-3 Advanced deployment solution topology

Due to the modular nature of these components, the enterprise can grow quickly and easily, providing cost savings and adapting to demands in a flexible way.

WebSphere Voice Response is configured so that it points to a single application that could require a multitude of topic or language resources. From the application, a WebSphere Voice Response calls one of many WebSphere Voice Server resources. For instance, the initial application could welcome the user and offer a choice of languages from which other applications may be linked.

The WebSphere Voice Response calls upon the appropriate WebSphere Voice Server to run the tasks, such as interpreting a caller's spoken input, which the WebSphere Voice Response, according to the application, might send that data to the WebSphere Application Server. From here, server scripts could communicate with other devices such as databases or authentication servers. Then the application might transfer the call to an agent or voicemail, or in the best case, provide the customer service needed.

Ultimately, the qualities and interconnectivity that WebSphere offers in a voice driven response environment enable infrastructure flexibility that is capable of meeting the needs of a growing enterprise.

Figure 2-4 shows an example of how infrastructure growth can occur across multiple interconnecting layers. Here three are languages supported. The languages for which we would expect lower call volumes can exist among Load Balancer clusters containing both ASR and TTS engines. If we expect a high call volume for a particular language, we may want to split the TTS and ASR engines into separate LB clusters. The Network Deployment cell is capable of offering management to all devices involved with the exception of the WebSphere Voice Response for AIX. The WebSphere Voice Response for AIX is managed by using the single system image (SSI) service.

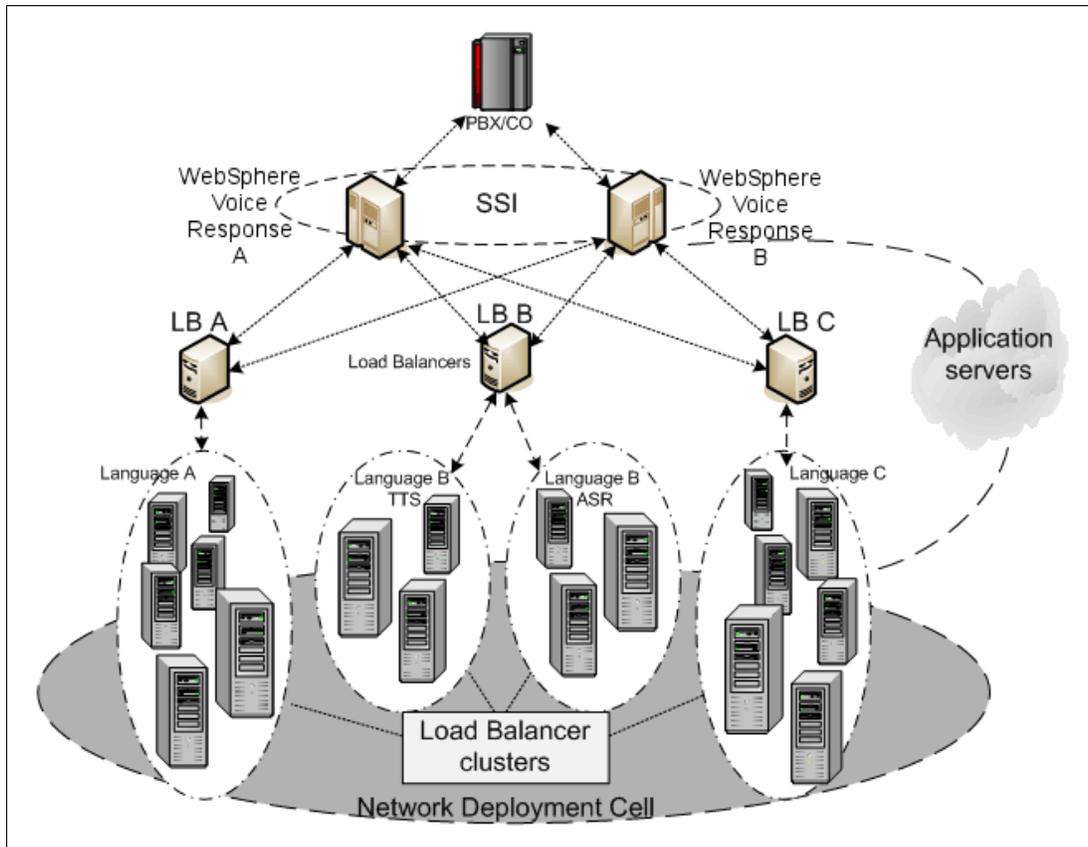


Figure 2-4 Advanced deployment solution topology with Load Balancer clusters and Network Deployment cell





## Requirements, installation, and configuration

In this chapter, we look at the hardware and software requirements to deploy both a basic and advanced configuration of WebSphere Voice Response system with WebSphere Voice Server. The examples that are presented are not intended, nor should they be used, as a requirements diagram for your particular deployment.

**Note:** Before deploying any solution, consult with your IBM specialist to create the architecture that best meets your needs. Then purchase the hardware and hardware configuration components, such as random access memory (RAM), disk storage, and number of processors.

This chapter also covers the installation and configuration of the IBM WebSphere Voice Server and WebSphere Voice Response in the test environment, at the IBM International Technical Support Organization (ITSO) lab, that was used to produce this Redpaper. It explains the steps that are required to install WebSphere Voice Response for AIX V4.2. And it describes the configuration for WebSphere Voice Response and WebSphere Voice Server to test their interoperability.

**WebSphere Voice Server for Multiplatforms Information Center:** Throughout this chapter, we refer you to the WebSphere Voice Server for Multiplatforms Information Center. You can find this information center on the Web at:

<http://publib.boulder.ibm.com/infocenter/pvcvoice/51x/index.jsp>

## 3.1 Requirements

This section describes the prerequisites of WebSphere Voice Server on Linux and the Windows Server 2003 platform.

### 3.1.1 Hardware requirements

Table 3-1 outlines the supported hardware for WebSphere Voice Server, which includes machines that meet the minimum hardware criteria as defined.

Table 3-1 Minimum supported hardware

Processor	Memory (RAM)	Free disk space	Network	Other
Intel Pentium® III processor, 1 GHz or equivalent processor	2 GB	2 GB	TCP/IP	CD

Table 3-2 presents the disk space requirements for the text-to-speech (TTS) server by voice.

Table 3-2 TTS server disk space requirements by voice

Language	WebSphere Voice Server (male voice)	WebSphere Voice Server (female or second voice)
Australian English	286 MB	264 MB
French Canadian	177 MB	292 MB
German	287 MB	339 MB
Japanese	63 MB	322 MB
Latin American Spanish	200 MB	360 MB
Simplified Chinese	N/A	253 MB
UK English	286 MB	264/276 MB
US English	275 MB	317/321 MB

Always refer to the WebSphere Voice Server for Multiplatforms Information Center for the latest hardware requirements:

<http://publib.boulder.ibm.com/infocenter/pvcvoice/51x/index.jsp>

### 3.1.2 LAN requirements

To handle the transport of voice data and control information between all WebSphere Voice Response and WebSphere Voice Server machines, place all these machines on a dedicated local area network (LAN). The minimum requirement is for a 100 Mbps full duplex Ethernet network or equivalent. The LAN technology used must be appropriate for the number of channels and machines.

It is difficult to precisely estimate the saturation point for a specific configuration. If you have questions about these issues, contact your IBM representative for advice on capacity planning.

### 3.1.3 Software requirements

WebSphere Voice Server V5.1.1 and V5.1.2 support the following operating systems:

- ▶ Red Hat Enterprise Linux WS/ES/AS V3.0 Update 1 only (2.4 Kernel) for Intel Pentium III processor (1 GHz or higher) or equivalent hardware
- ▶ SUSE Linux Enterprise Server V8, powered by United Linux V1.0 (SLES-8) with Service Pack 2a or 3 for Intel Pentium III processor (1 GHz or higher) or equivalent hardware
- ▶ Windows Server 2003 Enterprise or Standard editions

Always refer to the WebSphere Voice Server for Multiplatforms Information Center for the latest software requirements:

<http://publib.boulder.ibm.com/infocenter/pvcvoice/51x/index.jsp>

## 3.2 WebSphere Voice Server installation

To install WebSphere Voice Server, refer to the *IBM WebSphere Voice Server for Multiplatforms V5.1.1/V5.1.2 Handbook*, SG24-6447. This IBM Redbook provides step-by-step instructions for each operating system supported by WebSphere Voice Server.

In addition, WebSphere Voice Server provides a *Getting started with IBM WebSphere Voice Server for Multiplatforms Guide* on the base CD-ROM. This guide presents a checklist approach for installing the product quickly. You can find additional installation instructions in the WebSphere Voice Server Information Center.

## 3.3 Configuring WebSphere Voice Server for WebSphere Voice Response

All WebSphere Voice Response customers need to run a configuration script so that WebSphere Voice Server is configured correctly. Use these steps to configure each WebSphere Voice Server for use with WebSphere Voice Response.

1. Ensure that WebSphere Application Server is running on the WebSphere Voice Server machine.
2. Change the directory to the WebSphere Application Server installation path, such as `/opt/WebSphere/AppServer/bin` (the Linux default installation directory) or `C:/Program Files/WebSphere/AppServer/bin` (the Windows default installation directory).
3. Run the `configureWVR` file for your platform:
  - For Linux: `./configureWVR.sh`
  - For Windows: `configureWVR.bat`
4. Stop WebSphere Application Server and then restart it.

Example 3-1 shows the output from running the script on a Linux system.

*Example 3-1 Running configureWVR.sh on a Linux WebSphere Voice Server machine*

---

```
[root@wvsdev112 bin]# cd /opt/WebSphere/AppServer/bin
[root@wvsdev112 bin]# ./configureWVR.sh
WASX7209I: Connected to process "server1" on node wvsdev112 using SOAP connector; The type
of process is: UnManagedProcess
-----
- configureWVR - updates Voice Server config for WVR -
-----
configureWVRI: Found WVSAdminConfig service.
configureWVRI: Updating configuration values...
configureWVRI: Update completed.
configureWVRI: Done.
[root@wvsdev112 bin]#
```

---

## 3.4 Tuning of WebSphere Voice Server for WebSphere Voice Response

For the best and most current resource regarding tuning WebSphere Voice Server, refer to the WebSphere Voice Server Information Center and search under *tuning*.

## 3.5 Administration Console

The Administration Console is a service provided by the WebSphere Application Server and is accessible by a Java enabled Web browser. It is a great part of the WebSphere Application Server products. You should already be well acquainted with its capabilities, because it plays a part in the management of all the WebSphere Application Server based products. The Administration Console offers many options for system administrators to leverage.

Using a Java enabled Web browser, go to the following URL:

```
http://hostname:9090/admin
```

The *hostname* variable may be *localhost* if you are working directly on the WebSphere Voice Server machine, the Domain Name System (DNS) host name, or the IP address if you are accessing it remotely.

**Note:** In an advanced configuration where more than one WebSphere Voice Server is managed as part of a Deployment Manager cell, the host name is that of the Deployment Manager machine and not an individual WebSphere Voice Server node.

The WebSphere Voice Server Administrative Console Login panel (Figure 3-1) is displayed in your browser. Until security is enabled, a user ID is not required, and you can click **OK** to log in. After you log in, you can control the login with user passwords that you define from within the console.



Figure 3-1 Administrative Console Login panel

From the navigation panel, click **Voice System View** → **Voice Servers** as shown on the left in Figure 3-2.

This opens the Voice Servers panel (right side of Figure 3-2) within the Administration Console. For a basic configuration with only one WebSphere Voice Server, you see only a single voice server in the list. If you are using Deployment Manager, then you see one or more voice servers in the list. Click the voice server name link, which is **bc1srv9\_server1** in our configuration.



Figure 3-2 Administrative Console Voice Servers panel

The Runtime tab for the bc1srv9\_server1 voice server is displayed as shown in Figure 3-3. It shows a snapshot of the current voice server system status as well as the status of the automatic speech recognition (ASR) and TTS engines.

The screenshot shows the WebSphere Voice Server Administrative Console interface. The main content area is titled "bc1srv9\_server1" and contains three sections: "General Status", "ASR Status", and "TTS Status".

**General Status**

CPU	<input type="text" value="0%"/>	Shows current CPU utilization
Free Memory	861,620KB	Shows current free memory in system
WVS Supporting Components		Shows overall status of voice server's non-speech runtime components

**ASR Status**

ASR Engines		Shows overall status of speech recognition engines
Active Sessions	<input type="text" value="0 of 8"/>	Specifies active reco sessions out of total sessions
Language(s) installed	English (U.S.) French (Canadian)	Specifies recognition languages installed

**TTS Status**

TTS Engines		Shows overall status of text-to-speech engines
Active Sessions	<input type="text" value="0 of 4"/>	Specifies active text-to-speech sessions out of total sessions
Default voice settings	Andrew	Specifies the voice attributes used when text-to-speech requests have unspecified settings (such as gender and age). For more information, please reference <a href="#">Information Center</a>
Voice(s) installed <small>* Identifies voices currently available for use</small>	English (U.S.), female - Allison * English (U.S.), male - Andrew * French (Canadian), female -	Specifies text-to-speech voice(s) and language installed. To make a voice available for use, please reference <a href="#">Information Center</a>

Figure 3-3 Administrative Console Voice Server Runtime panel

As shown in Figure 3-4, from the Configuration tab for a selected server, you can start and stop servers, as well as and assign the number of engines to be allocated to the TTS and ASR components. If you enter the URI of a grammar or lexicon file into the Grammar pre-loads or Lexicon pre-loads fields respectively, then that file is preloaded to ensure rapid response. This is especially important when you have large grammars that need to be parsed. These files are held in a cache whose timeout and size are configured from the `$WAS_ROOT/bin/wsadmin.sh` WebSphere Application Server configuration script.

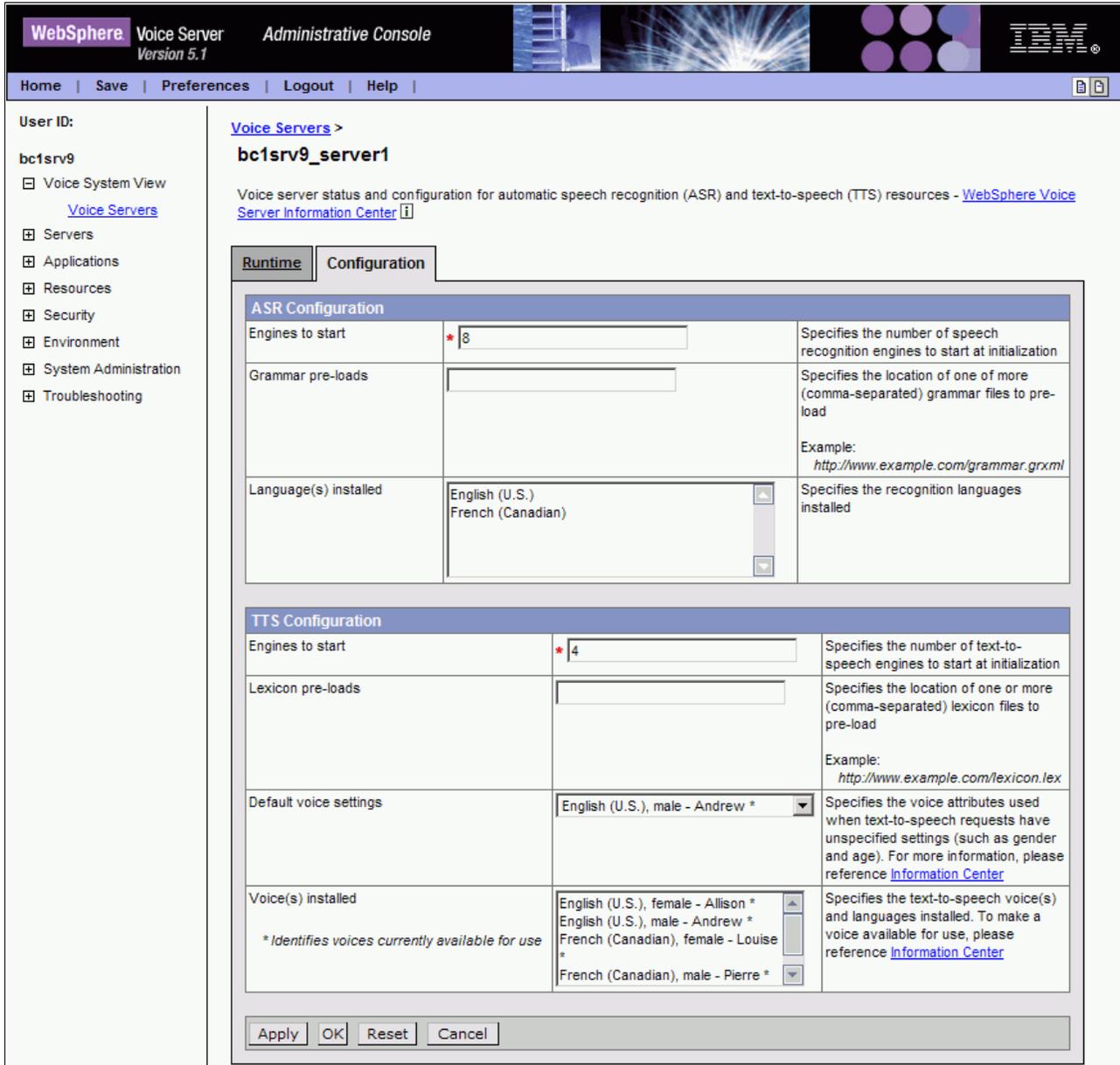


Figure 3-4 Administrative Console Voice Server Configuration panel

Alerts shown in the bottom frame of the Administration Console (Figure 3-5) can point to problem areas due to misconfigurations indicating which elements should be traced.

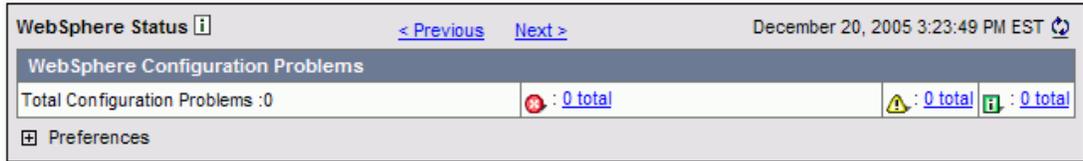


Figure 3-5 Administrative Console panel showing WebSphere configuration problems

To enable logging, click **Troubleshooting** → **Logs and Trace** and then click the name of the voice server for which you want to enable tracing. The Diagnostic Trace Service Configuration panel is displayed. To enable the trace log attributes from the Runtime tab, click **Runtime**. Then you see results similar to those shown in Figure 3-6.

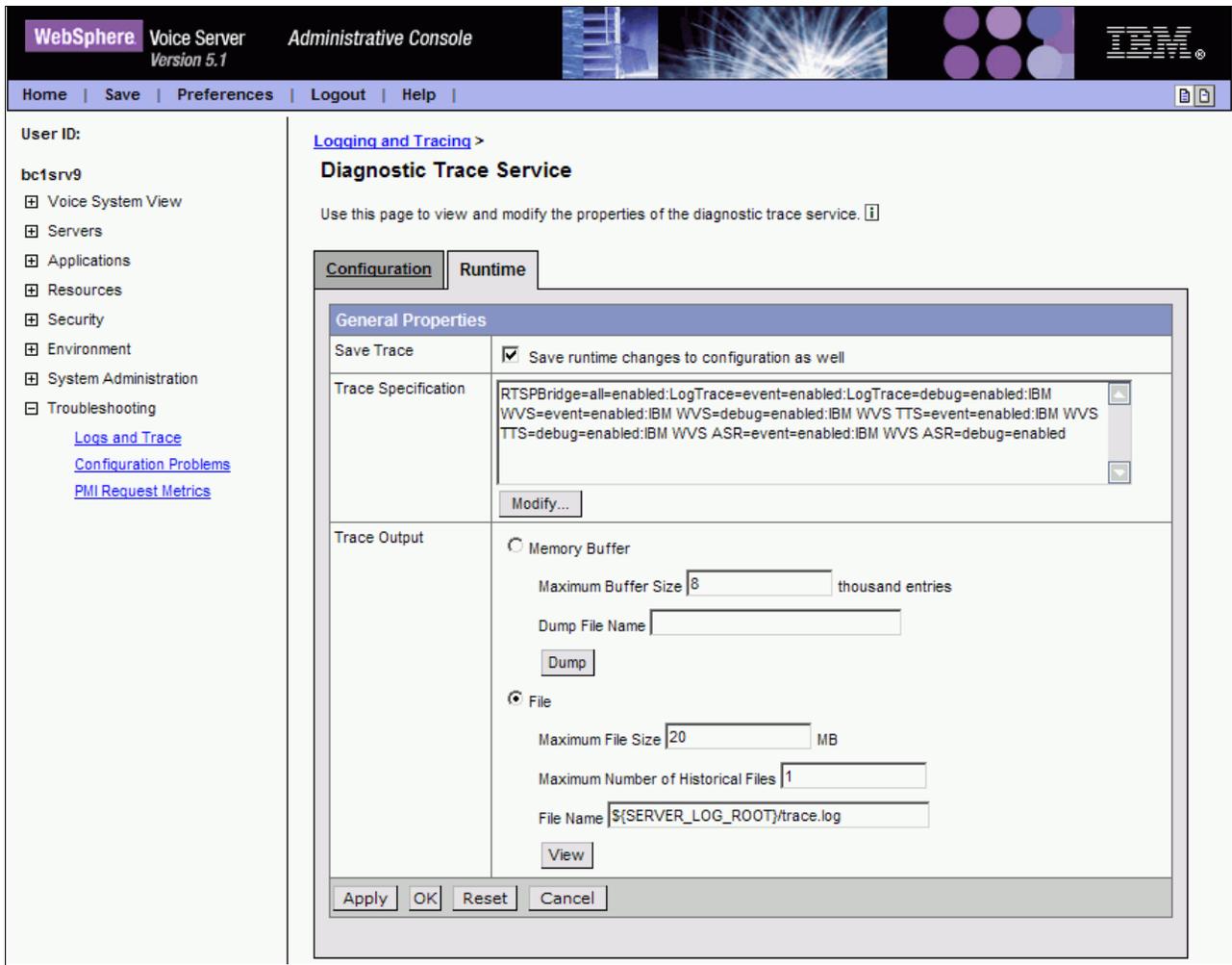


Figure 3-6 Administrative Console Diagnostic Trace Service Runtime panel to enable logging

When the trace log attributes are enabled, the logs can show real-time events immediately. You can define the name and location of the trace results from the same panel. To enter log attributes, click **Modify...** to open a list of choices. A window opens (see Figure 3-7) in the Groups and Components tabs.

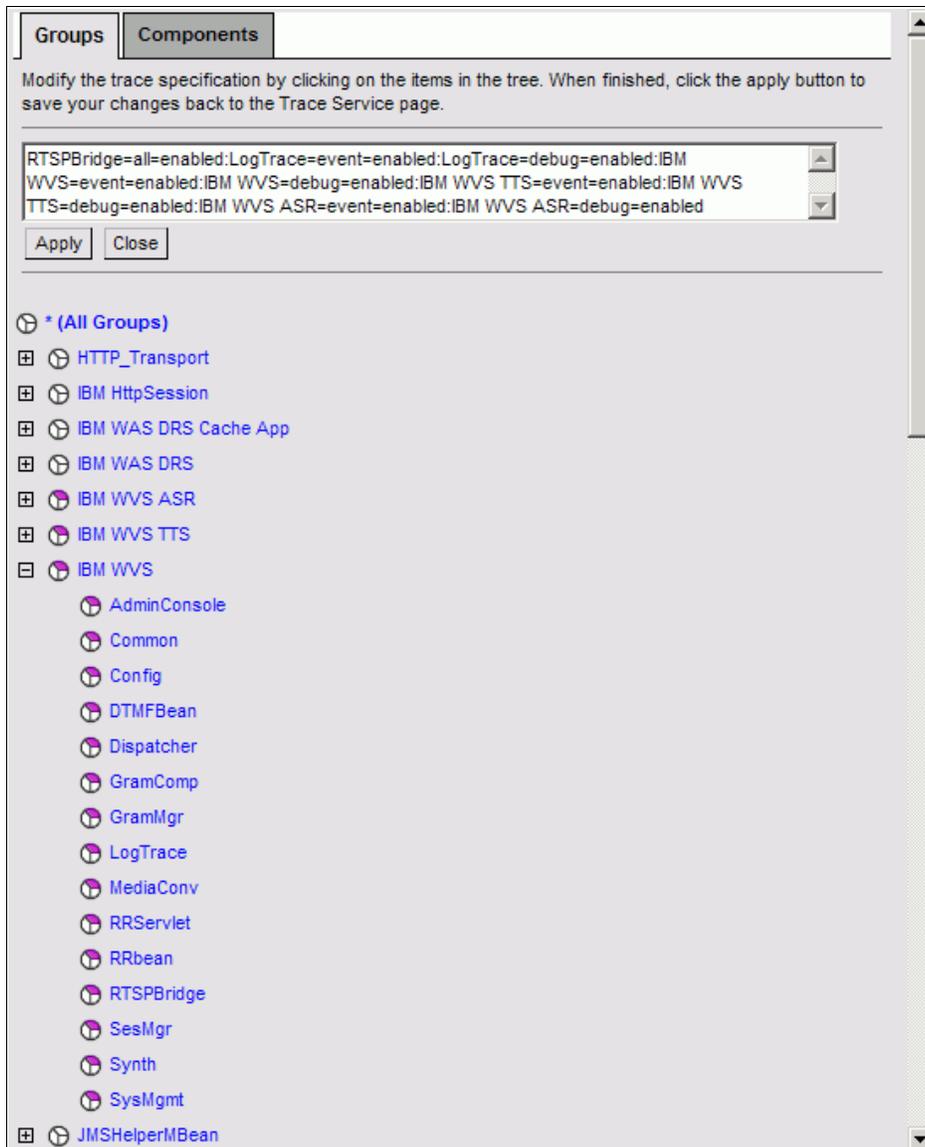


Figure 3-7 Administrative Console traceable items

Choose an item to trace by clicking its name from the tree. A small pop-up menu, like the example shown in Figure 3-8, is displayed that shows the trace specifications for that item.



Figure 3-8 Administrative Console trace specifications

When you are finished enabling or disabling the various trace specifications, click **Apply**. You must save the configuration and restart the server for the changes to take effect. With the settings enabled, tracing will be vital to debugging the applications and WebSphere Voice Response configuration issues.

The features of Administration Console are evident when used from the Deployment Manager to monitor multiple remote servers, yet they may lack some details that the command line scripts offers. You can only run the command line scripts locally, which restricts actions to the local servers and should be used for initial configuration.

Command line scripts offer a powerful interface for maintaining a WebSphere Voice Server system. In the `/opt/WebSphere/VoiceServer/samples` directory, many shell scripts can be used, all of which have option commands that can be found by entering this command:

```
./<script> --help
```

When deploying WebSphere Voice Server to serve WebSphere Voice Response, little needs to be done because WebSphere Voice Server and WebSphere Voice Response work together to communicate. WebSphere Voice Server is ready to reply to any appropriate request. After you define the ASR and TTS resources in the `default.cff` file for the WebSphere Voice Response system, the two systems simply start talking to each other.

If there is an error, check that all the components are properly installed and that the servers restarted. The majority of errors that may occur in this scenario are from within the voice application or the WebSphere Voice Response configuration.

## 3.6 Installation of WebSphere Voice Response

This section examines the features and functions of WebSphere Voice Response for AIX V4.2. In addition, it looks at WebSphere Voice Server Connector, which makes WebSphere Voice Response interoperable with WebSphere Voice Server V5.1.x.

### 3.6.1 WebSphere Voice Response for AIX V4.2

WebSphere Voice Response for AIX V4.2 delivers the interactive voice response (IVR) system of choice for medium and large enterprises. WebSphere Voice Response for AIX V4.2 retains most of the features in earlier versions.

In addition, WebSphere Voice Response for AIX V4.2 offers several enhanced features in this version.

- ▶ Enhanced Telephony Protocol support for Voice over IP (VoIP) telephony via the Session Initiation Protocol (SIP)
- ▶ Service provider networks: SS7 T1 (ANSI Signalling System #7)
- ▶ Enterprise networks: QSig, R2MFC Korean
- ▶ Call Control Extensible Markup Language (CCXML) V1.0 support
- ▶ Enhanced VoiceXtensible Markup Language (VoiceXML) V2.0 support
- ▶ Scalability to 480 channels per system
- ▶ Voice Toolkit for WebSphere Studio
- ▶ Interoperability with WebSphere Voice Server V4.2: For speech recognition and TTS applications
- ▶ Improved echo cancellation for speech recognition-enabled applications
- ▶ Interoperability with WebSphere Voice Application Access V5.0 for Voice Portal applications
- ▶ Migration from WebSphere Voice Response for AIX V3.1
- ▶ New fax card compatible with the latest pSeries machines
- ▶ Fully accessible WebSphere Voice Response when running CCXML V2.0 and VoiceXML V2.0 applications
- ▶ Enhanced support of Unified Messaging for WebSphere Voice Response V4.2

For a complete list of WebSphere Voice Response for AIX V4.2 functions and other IBM voice related products, refer to the IBM WebSphere Voice Web site:

<http://www.ibm.com/developerworks/websphere/zones/voice>

The key enhancement of WebSphere Voice Response for AIX V4.2 is the support of WebSphere Voice Server V5.1.x using the WebSphere Voice Server Connector.

## Hardware requirements

The hardware requirements for WebSphere Voice Response for AIX V4.2 include the following items:

- ▶ Selected models of IBM RS/6000® or pSeries system with the appropriate amount of RAM and hard disk space
- ▶ IBM Quad Digital Trunk X Adapter (DTXA), Digital Trunk Telephony Adapter (DTTA), or Digital Trunk Extended Adapter (DTEA) adapter to provide the connectivity to the telephony network

## Software requirements

The mandatory software requirement is IBM AIX V5.2 with minimum Maintenance Level 2 or later. Depending on your business requirements, some of the chargeable optional software available are:

- ▶ Communication Server for AIX to support 3270 emulation
- ▶ Single system image (SSI) for multi WebSphere Voice Response for AIX V4.2 systems
- ▶ High Availability Cluster Multi-Processing (HACMP™) V5.1 for high availability and redundancy

- ▶ IBM Java Development Kit for VoiceXML, CCXML, and Java application programming interface (API)
- ▶ VisualAge C++ V6.0 compiler for custom server development

Choosing the correct hardware and software combination to meet your business requirements is part of the planning process in the implementation of WebSphere Voice Response for AIX V4.2. For more information and support, contact an IBM representative.

### 3.6.2 WebSphere Voice Response for AIX V4.2 plug-ins

As part of the WebSphere Voice Response for AIX V4.2 installation (which we explain later), you need to install the dtjmrpc.zip plug-in. Meanwhile, assuming that WebSphere Voice Response for AIX V4.2 is successfully installed, two custom servers and four state tables are generated as shown in Figure 3-23 on page 50 and Figure 3-24 on page 50. The custom servers and state tables provide the necessary links for WebSphere Voice Response for AIX V4.2 to communicate with WebSphere Voice Server V5.1.x using the WebSphere Voice Server Connector. To use these links, you need to define the plug-in class correctly.

You must correctly define the plug-in class to support WebSphere Voice Server in the default.cff file by using RecoService and TTSService keywords. Example 3-2 shows the definition used to identify the speech recognition service provided by WebSphere Voice Server V5.1.x.

*Example 3-2 RecoService entry in default.cff for WebSphere Voice Server V5.1.x*

---

```
RecoService=Reco_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPreco
  InitSessionString=URI=rtsp://9.42.171.98/media/recognizer
  RecoType=Recoen_US
```

---

Example 3-3 shows the definition for a TTS service.

*Example 3-3 TTSService entry in default.cff for WebSphere Voice Server V5.1.x*

---

```
TTSService=Tts_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPTTS
  InitSessionString=URI=rtsp://9.42.171.99/media/synthesizer
  TTSType=Ttsen_US
```

---

Other plug-in classes are available to perform different functions. The dtjvv.zip plug-in is available for use to support the IBM WebSphere Voice Server V4.2 speech technology.

## 3.7 ITSO WebSphere Voice Response installation

This section describes the installation of WebSphere Voice Response for AIX V4.2 on the IBM @server pSeries 7028-6E4 server and presents the steps used to verify that the installation completed successfully. It also explains the deployment of some sample VoiceXML applications in the ITSO laboratory environment.

### 3.7.1 Interconnection to voice network

With WebSphere Voice Response for AIX V4.2, you can choose from a rich selection of time division multiplexing (TDM) T1/E1 protocols for interconnection to PABXs, telecommunication central office (CO) switches, or both. It also enables you to choose to interconnect it to an IP network for Voice over IP (VoIP) deployment. If you intend to deploy WebSphere Voice Response for AIX V4.2 in the VoIP environment, you have to install the DTEA adapter and its associated software on a suitable pSeries server.

In the ITSO laboratory, we used the Gordon Kapes System 930 Telephony Simulator to emulate a North American telecommunication central office (CO) telephone exchange. We attached the following items to it:

- ▶ Two analog telephones

These telephones were used to make calls to WebSphere Voice Response for AIX V4.2.

- ▶ Two T1 links

The first T1 link was used on a different project. We connected the second T1 link to the DTXA adapter in the pSeries server. It emulated the ISDN\_5E11-5E15 National 2 network side of the Integrated Services Digital Network (ISDN) protocol. Channel 24 of the T1 link is reserved for ISDN common channel signaling. This reduces the number of available channels to 23.

To make test calls to WebSphere Voice Response for AIX V4.2 from analog telephones, we dialed the digit 9 plus a seven-digit number, such as 789 4563. The digit 9 was defined in the Gordon Kapes System 930 Telephony Simulator configuration as a way to access one of the T1 lines. The seven-digit number will appear as called number to the WebSphere Voice Response for AIX V4.2. The dial plan of the Gordon Kapes System was configured to emulate the American numbering plan.

To directly select a particular channel in the second T1 link, we dialed \*2 (asterisk and 2, to select the second T1 link), a two-digit value of 01 to 23 (to select a channel in the T1 link), plus a seven-digit number. For example, to select the second T1 line and use channel 01 to call 7894563, we dialed the following number:

```
*2 01 789 4563
```

This method of dialing allows us to individually test all the channels in the T1 link after installing WebSphere Voice Response for AIX V4.2. The seven-digit number will appear as the called number to the WebSphere Voice Response for AIX V4.2.

### 3.7.2 ITSO setup

Figure 3-9 illustrates the interconnection between the pSeries running WebSphere Voice Response for AIX V4.2 and the Gordon Kapes System 930 Telephony simulator. It is the same setup that we used in the ITSO laboratory. Not shown is the LAN connection between the pSeries and the Load Balancer, Deployment Manager, and WebSphere Application Server.

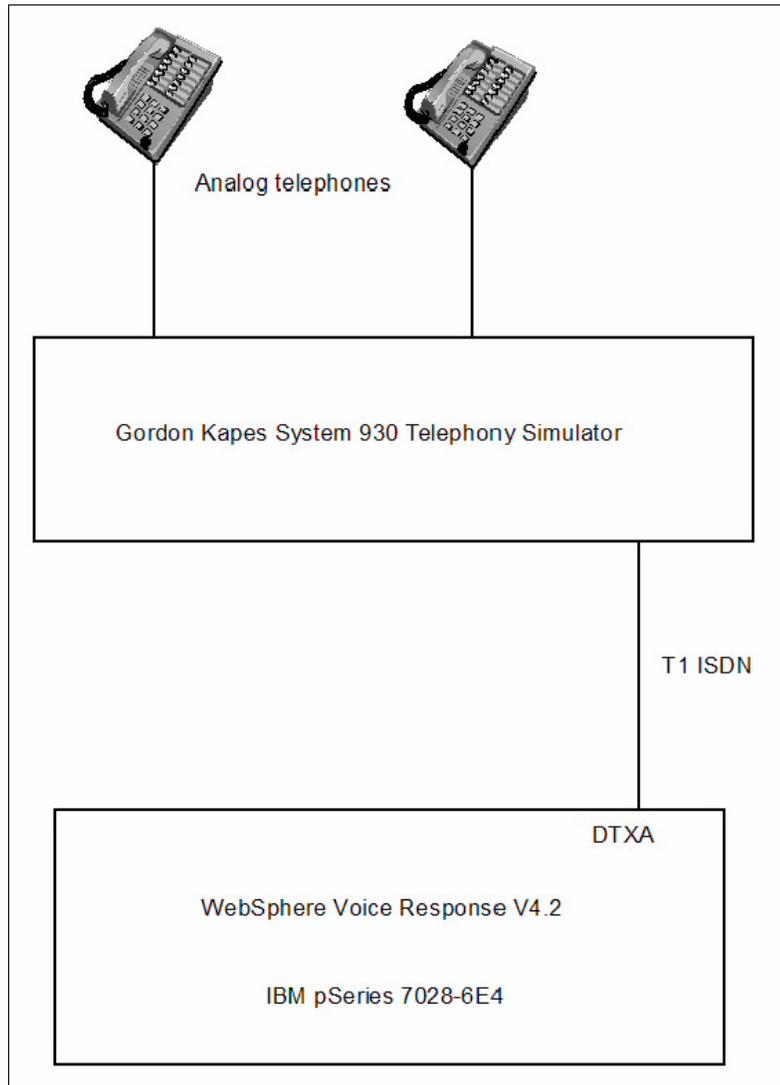


Figure 3-9 WebSphere Voice Response for AIX V4.2 system and telephony simulator interconnection

The setup allows us to use telephones to make calls to WebSphere Voice Response for AIX V4.2, which either fetches a local VoiceXML document stored on the pSeries server or fetches a VoiceXML document from the HTTP Web server. When the VoiceXML document is executed, the WebSphere Voice Server V5.1.x is called to perform the necessary TTS and speech recognition functions.

### 3.7.3 IBM eServer pSeries 7028-6E4

Before we installed WebSphere Voice Response for AIX V4.2, we verified that the pSeries 7028-6E4 had the hardware and software capabilities to support WebSphere Voice Response for AIX V4.2. Since AIX V5.2 was pre-installed on the pSeries, we could use some of the AIX commands to examine the hardware and software components installed on the machine. We used the `oslevel -r` command to obtain the operating system maintenance level. The response to the command was 5200-02, which met the requirement to run WebSphere Voice Response for AIX V4.2.

We used the `lsdev -C` command to view the hardware configuration within the pSeries system. Example 3-4 shows the output indicating that the machine is equipped with the DTXA adapter. It is the essential piece of hardware that is required to support the WebSphere Voice Response for AIX V4.2, and interconnected the pSeries system with the voice network. In the ITSO installation, we connected the DTXA adapter to the Gordon Kapes System 930 Telephony Simulator.

*Example 3-4 Partial output of the `lsdev -C` command showing the installed DTXA adapter*

---

ddriciop0	Available	11-09-00	IBM ARTIC960RxD PCI Device Driver
dtdd	Available		DirectTalk system-wide Telephony Kernel Extension
dtline0	Available	11-09-01	Artic card currently owned by Voice Response
dtpack0_0	Available		DirectTalk Digital Trunk Processor.
dtpack0_1	Available		DirectTalk Digital Trunk Processor.
dtpack0_2	Available		DirectTalk Digital Trunk Processor.
dtpack0_3	Available		DirectTalk Digital Trunk Processor.
dtxa0	Available		Voice Response Digital Trunk X Adapter
riciop0	Available	11-09	IBM ARTIC960RxD Quad Digital Trunk PCI Adapter

---

As illustrated in Example 3-5, two SCSI hard drives of 32 GB each were installed, which are sufficient to support WebSphere Voice Response for AIX V4.2.

*Example 3-5 Details of the two SCSI drives installed in the pSeries server*

---

```
$ lspv hdisk0
PHYSICAL VOLUME:      hdisk0                VOLUME GROUP:      rootvg
PV IDENTIFIER:        0006957f623a5c50 VG IDENTIFIER      000695bf00004c00000001028
725f3a4
PV STATE:              active
STALE PARTITIONS:     0                ALLOCATABLE:       yes
PP SIZE:               64 megabyte(s)        LOGICAL VOLUMES:   11
TOTAL PPs:             542 (34688 megabytes)  VG DESCRIPTORS:    2
FREE PPs:              0 (0 megabytes)        HOT SPARE:         no
USED PPs:              542 (34688 megabytes)
FREE DISTRIBUTION:    00..00..00..00..00
USED DISTRIBUTION:   109..108..108..108..109

$ lspv hdisk1
PHYSICAL VOLUME:      hdisk1                VOLUME GROUP:      rootvg
PV IDENTIFIER:        000695bf46770503 VG IDENTIFIER      000695bf00004c00000001028
725f3a4
PV STATE:              active
STALE PARTITIONS:     0                ALLOCATABLE:       yes
PP SIZE:               64 megabyte(s)        LOGICAL VOLUMES:   1
TOTAL PPs:             542 (34688 megabytes)  VG DESCRIPTORS:    1
FREE PPs:              0 (0 megabytes)        HOT SPARE:         no
USED PPs:              542 (34688 megabytes)
FREE DISTRIBUTION:    00..00..00..00..00
USED DISTRIBUTION:   109..108..108..108..109
```

---

Because the memory size is not available from the output of `lsdev -C` command, we used the `lsattr` command instead as shown in Example 3-6. The output of the command shows that the pSeries server is equipped with 16 GB of RAM.

*Example 3-6 Memory size in the pSeries server*

---

```
# lsattr -HE -l sys0 -a realmem
attribute value      description                               user_settable
realmem    16777216 Amount of usable physical memory in Kbytes False#
```

---

We changed the default AIX configuration to not start the Common Desktop Environment (CDE) at boot time. We ran WebSphere Voice Response for AIX V4.2 on the AIX windows environment instead of CDE. This reduced the amount of processing overhead. No AIX system tuning parameters were changed in the ITSO implementation because our applications do not justify the changes. However, in a full production system, you may have to tune some of the operating system parameters.

We made all the changes to the AIX system parameters using the System Management Interface Tool (SMIT). SMIT generates two log files, `smit.script` and `smit.log`. These files provide an audit log of the configuration changes that have been made. We recommend that you use SMIT to perform any changes to AIX and to install WebSphere Voice Response for AIX V4.2.

We used the electronic software download (ESD) CD images to install WebSphere Voice Response for AIX V4.2. For your installation, you most likely will install directly from the CDs. The ESD CD images that we used have the same content as the actual CDs.

### **pSeries hardware and software**

The following hardware configuration was installed on the pSeries 7028-6E4:

- ▶ Two 32 GB hard drives
- ▶ 16 GB of RAM
- ▶ DTXA adapter

The following software components were installed:

- ▶ AIX V5.2
- ▶ AIX Java Runtime V1.4.1
- ▶ License Use Management V5.2
- ▶ WebSphere Voice Response for AIX V4.2

## **3.7.4 Installing WebSphere Voice Response for AIX V4.2**

Before we started our installation, we checked that the pSeries 7028-6E4 can support the DTXA adapter. This is essential because not all pSeries servers can support the DTXA. We installed the DTXA adapter and connected it to the telephony simulator with a shielded twisted-pair cable. Because the distance between the equipment is about three meters, the shielded twisted-pair cable works well.

WebSphere Voice Response for AIX V4.2 manages its own database using IBM DB2 Universal Database™ V8.1.1. This is supplied with WebSphere Voice Response for AIX V4.2. If you have your own instances of DB2 running on the machine where you are going to install WebSphere Voice Response for AIX V4.2, make sure that your instance of DB2 does not interfere with the WebSphere Voice Response for AIX V4.2 database.

During the installation of WebSphere Voice Response for AIX V4.2, two AIX users are automatically created:

- ▶ dtuser
- ▶ dtdb23in

You log in as dtuser to start, stop, and manage WebSphere Voice Response for AIX V4.2. The *dtdb23in* user is implicitly used by WebSphere Voice Response for AIX V4.2 to manage its database. In normal operating conditions, you do not need to login as dtdb23in.

Depending on the WebSphere Voice Response for AIX V4.2 options you are installing, more AIX users can be created. You must read the accompanying documentation to find the name of the AIX user that will be created automatically. By coincidence, the newly created name might match the name of an existing AIX user.

The installation of WebSphere Voice Response for AIX V4.2 is carried out with SMITTY by using the following steps:

1. Log in as root. Because AIX V5.2 was preloaded, check that it is at the required level to support WebSphere Voice Response for AIX V4.2 by using the `oslevel -r AIX` command. In addition, type the `lspp` command with different arguments (see Table 3-3) to ensure that the desired AIX file sets are installed.

*Table 3-3 lspp commands to check the installed file sets*

Command	Purpose
<code>lspp -L "bos.*"   more</code>	To verify that desired bos file sets are installed
<code>lspp -L "X11*"   more</code>	To verify that desired AIX windows are installed
<code>lspp -L "i for*"   more</code>	To verify that License Use Management file sets are installed

2. Verify that the correct version of Java was installed by using the `java -version` command. The command output shown in Example 3-7 indicates that the correct version of Java is installed.

*Example 3-7 The java -version command shows J2RE V1.4.1 installed*

```
$ java -version
java version "1.4.1"
Java(TM) 2 Runtime Environment, Standard Edition (build 1.4.1)
Classic VM (build 1.4.1, J2RE 1.4.1 IBM AIX build ca1411-20040301 (JIT enabled:
jitc))
```

3. Type the following command to start the installation of WebSphere Voice Response for AIX V4.2:
 

```
smitty install_latest
```
4. Select the relevant file sets and the options that you purchased. If the installation fails, check the error messages and then take the appropriate actions.
5. When the installation is successfully completed, two AIX users are created, dtuser and dtdb23in. Use the `smitty users` command to change the password of dtuser. There is no need to change the password of dtdb23in.
6. Assign dtuser as the owner of WebSphere Voice Response base file sets. Ignore the Structured Query Language (SQL) error messages because the database has not yet been created.

7. Install WebSphere Voice Response for AIX V4.2 updates by using the following command:

```
smitty update_all
```

We selected the **COMMIT software updates** option at this stage since there is not much software to recover if there are problems with the installed software.

8. After you apply the program temporary fix (PTF) from the previous step, assign dtuser as the owner of updated WebSphere Voice Response file sets. Then you see a response similar to the one in Example 3-8.

*Example 3-8 Setting dtuser as owner of WebSphere Voice Response file sets*

---

```
# /usr/lpp/dirTalk/tools/vae.setuser
vae.setuser: Tue May 31 17:52:35 EDT 2005 - Starting.
ssistatus: Querying configuration ..
ssistatus: m10695bf is a Standalone system

vae.setuser: Enter user ID for running WebSphere Voice Response.
              (Must begin with alpha character): dtuser

vae.setuser: Process All filesystems? ([y]/n/ or ? for help): y
vae.setuser: Processing ALL mounted filesystems.

vae.setuser: Setting file ownership and permissions using AIX fileset inventory.
vae.setuser: Checked fileset dirTalk.DT.db
vae.setuser: Checked fileset dirTalk.DT.dms
vae.setuser: Checked fileset dirTalk.DT.rte
vae.setuser: Checked fileset dirTalk.DT.sample
vae.setuser: Checked fileset dirTalk.DT.unicode
vae.setuser: Checked fileset dirTalk.DT.help
vae.setuser: Checked fileset dirTalk.ISDN.ATT
vae.setuser: Checked fileset dirTalk.ISDN.DMS100
vae.setuser: Checked fileset dirTalk.ISDN.Euro-ISDN
vae.setuser: Checked fileset dirTalk.ISDN.INS1500
vae.setuser: Checked fileset dirTalk.ISDN.com
vae.setuser: Checked fileset dirTalk.SP.de_DE
vae.setuser: Checked fileset dirTalk.SP.en_GB
vae.setuser: Checked fileset dirTalk.SP.fr_FR
vae.setuser: Checked fileset dirTalk.SP.it_IT
vae.setuser: Checked fileset dirTalk.SP.nl_BE
vae.setuser: Checked fileset dirTalk.SP.pt_BR
vae.setuser: Checked fileset dirTalk.TDD
vae.setuser: Checked fileset dirTalk.VOIP_SIP
vae.setuser: Checked fileset dirTalk.VRBE_XML.VRnode.rte
vae.setuser: Checked fileset dirTalk.VRBE_XML.VRnode.samples
vae.setuser: Checked fileset dirTalk.VRBE_XML.ca_ES
vae.setuser: Checked fileset dirTalk.VRBE_XML.de_DE
vae.setuser: Checked fileset dirTalk.VRBE_XML.en_GB
vae.setuser: Checked fileset dirTalk.VRBE_XML.en_US
vae.setuser: Checked fileset dirTalk.VRBE_XML.es_ES
vae.setuser: Checked fileset dirTalk.VRBE_XML.es_MX
vae.setuser: Checked fileset dirTalk.VRBE_XML.fr_CA
vae.setuser: Checked fileset dirTalk.VRBE_XML.fr_FR
vae.setuser: Checked fileset dirTalk.VRBE_XML.it_IT
vae.setuser: Checked fileset dirTalk.VRBE_XML.rte
vae.setuser: Running "lppchk -f" for verification.
vae.setuser: Setting ownership of directories. user_id=dtuser, group=staff.
vae.setuser: Setting ownership of files..... user_id=dtuser, group=staff.
vae.setuser: Tue May 31 17:56:20 EDT 2005 - Completed.
```

---

9. Restart AIX by typing the following command:

```
shutdown -rF
```

10. Log in as root. Type the following command to set the environment as though you are dtuser. Note that there is a space between the period and the forward slash.

```
. /usr/lpp/dirTalk/tools/vae.setenv
```

Confirm that the environment is correctly set to /usr/lpp/dirTalk by typing:

```
echo $VAE
```

11. Create the database.

a. Type the following command to check that the /home and /tmp file systems have 80 MB and 40 MB respectively of free disk available:

```
df -k
```

If the file systems do not have enough free disk space, increase the disk space by typing the following command:

```
smitty storage
```

b. Type the following command:

```
/usr/lpp/dirTalk/tools/DTdatabase -r
```

c. When you see the prompt "Restore supplied sample database and voice data? (y/n)", type y for yes.

Then the database is created. Example 3-9 shows the response for the ITSO installation.

*Example 3-9 Messages as database is created*

---

```
# /usr/lpp/dirTalk/tools/DTdatabase -r
```

```
DTdatabase: Starting 31-May-2005 at 18:46:18 [20504]
```

```
DTdatabase: Problem diagnosis output is saved in /tmp/DTdatabase.out.20504
```

```
DTdatabase: Restore supplied sample database and voice data? (y/n): y
```

```
DTdatabase: Restore starting...
```

```
DTdatabase: Uncompressing sample DB2 database image...
```

```
DTdatabase: Starting WVR DB2 instance...
```

```
DTdatabase: Forcing all apps to disconnect from WVR DB...
```

```
DTdatabase: Restoring DB2 image for database DTDBV230...
```

```
DTdatabase: Starting WVR DB2 instance...
```

```
DTdatabase: Increasing the Database Application Heap Size.
```

```
DTdatabase: Forcing all apps to disconnect from WVR DB...
```

```
DTdatabase: Stopping WVR DB2 instance...
```

```
DTdatabase: Starting WVR DB instance...
```

```
DTdatabase: Add help data to database DTDBV230...
```

```
DTdatabase: Removing existing voice data...
```

```
DTdatabase: This may take many minutes if you have a lot of voice stored ...
```

```
DTdatabase: Restoring default voice data...
```

```
fsupdate: Preparing to update filesystem information
```

```
fsupdate: Querying current configuration
```

```
fsupdate: Storing local WebSphere Voice Response user information
```

```
fsupdate: Creating Filesystem Identification Files
```

```
fsupdate: Completed creation of identification files
```

```
fsupdate: Ref is
```

```
"00000000-000010be-429ce998:00000000-000010bf-429ce998:00000000-000010c0-429ce998:00000000-000010c1-429ce998:00000000-000010c2-429ce998:"
```

```
fsupdate: Filesystem checksum successfully updated
```

DTdatabase: Removing Custom Server data...  
DTdatabase: Completed successfully 31-May-2005 at 18:47:53

#

12. Verify that the root environment is still /usr/lpp/dirTalk by typing the following command:

```
echo $VAE
```

If the environment is not as indicated, repeat step 10.

Example 3-10 shows that the DTXA card is installed. The last digit of riciop0 is the adapter number. In this case, it is zero.

*Example 3-10 DTXA adapter installed*

---

```
$ lsdev -C | grep riciop
ddriciop0 Available 11-09-00 IBM ARTIC960RxD PCI Device Driver
riciop0 Available 11-09 IBM ARTIC960RxD Quad Digital Trunk PCI Adapter
```

---

13. Assign dtuser as the owner of the DTXA adapter by using the following commands:

```
dt_setowner -s0
dt_setowner: riciop0 (DTXA) is now owned by WebSphere Voice Response (dtline0)
```

Reconfirm that the DTXA ownership was successfully assigned to dtuser by using the following command (see Example 3-4 on page 31):

```
lsdev -C
```

14. WebSphere Voice Response for AIX V4.2 is license-enabled software and is enrolled and managed by the IBM License Use Management. Example 3-11 shows the version of License Use Management that is used.

*Example 3-11 IBM License Use Management file sets in the ITSO installation*

---

```
# ls1pp -al "ifor*"
Fileset                                Level  State      Description
-----
Path: /usr/lib/objrepos
ifor_ls.base.cli                       5.2.0.30 COMMITTED License Use Management Runtime
                                         Code
ifor_ls.java.gui                       5.2.0.11 COMMITTED License Use Management Runtime
                                         Java GUI
ifor_ls.msg.en_US.base.cli             5.2.0.30 COMMITTED LUM Runtime Code Messages -
                                         U.S. English
ifor_ls.msg.en_US.java.gui             5.2.0.11 COMMITTED LUM Java Gui Messages - U.S.
                                         English

Path: /etc/objrepos
ifor_ls.base.cli                       5.2.0.30 COMMITTED License Use Management Runtime
                                         Code
```

---

We configured the pSeries as the license server. We added the following path to the ".profile":

```
PATH=/usr/opt/ifor/ls/os/aix/gin:$PATH
```

15. While still logged in as root, stop the License Use Management runtime with the **i4cfg -stop** command. Example 3-12 shows License Use Management processes 0513-044 (subsystems i4gdb and i4lmd) were successfully stopped. If you issue the command again, the response no longer shows the processes.

*Example 3-12 Stopping License Use Management*

---

```
# i4cfg -stop
i4cfg Version 4.6.7 AIX -- LUM Configuration Tool
(c) Copyright 1995-2003, IBM Corporation, All Rights Reserved
US Government Users Restricted Rights - Use, duplication or disclosure
restricted by GSA ADP Schedule Contract with IBM Corp.

0513-044 The i4gdb Subsystem was requested to stop.
0513-044 The i4lmd Subsystem was requested to stop.
```

---

16. Start the License Use Management configuration tool with the **i4cfg** command. Example 3-13 shows that License Use Management subsystems i4lmd and i4gdb were successfully started. If you want to run License Use Management on another host, you need to export the display to your designated host with using the following command, where *ip\_address* is the IP address of the designated host:

```
export=ip_address:0
```

*Example 3-13 Starting License Use Management*

---

```
# i4cfg -start
i4cfg Version 4.6.7 AIX -- LUM Configuration Tool
(c) Copyright 1995-2003, IBM Corporation, All Rights Reserved
US Government Users Restricted Rights - Use, duplication or disclosure
restricted by GSA ADP Schedule Contract with IBM Corp.

0513-059 The i4lmd Subsystem has been started. Subsystem PID is 47608.
0513-059 The i4gdb Subsystem has been started. Subsystem PID is 66640.

'Start Services' has completed successfully
```

---

17. The License Use Management configuration tool window (Figure 3-10) opens. In our ITSO installation, we used the pSeries as the Network License Server (NetworkLS) and as the Central Registry License Server (CrLS). We selected these boxes and the mandatory Network License Client check box.



Figure 3-10 License Use Management Configuration Tool window

18. Because the host name of the pSeries is m10695bf and it performs the NetworkLS and CrLS functions, we add this host name to the Servers field, as shown in the top of Figure 3-11.

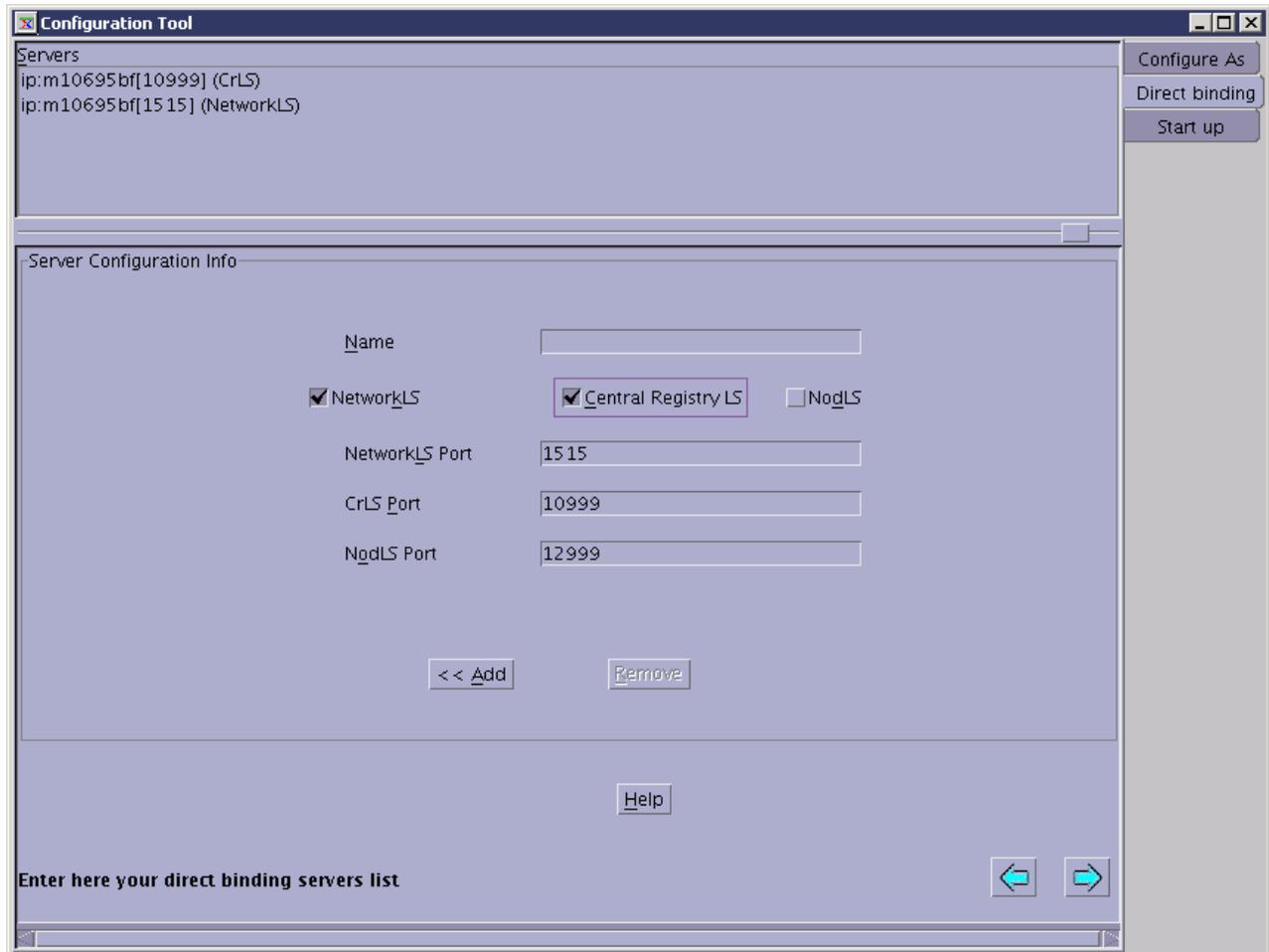


Figure 3-11 License Use Management Direct Binding window with the pSeries functions as NetworkLS and CrLS

19. The server is required to start License Use Management services at system start up. We select that option as shown in Figure 3-12.

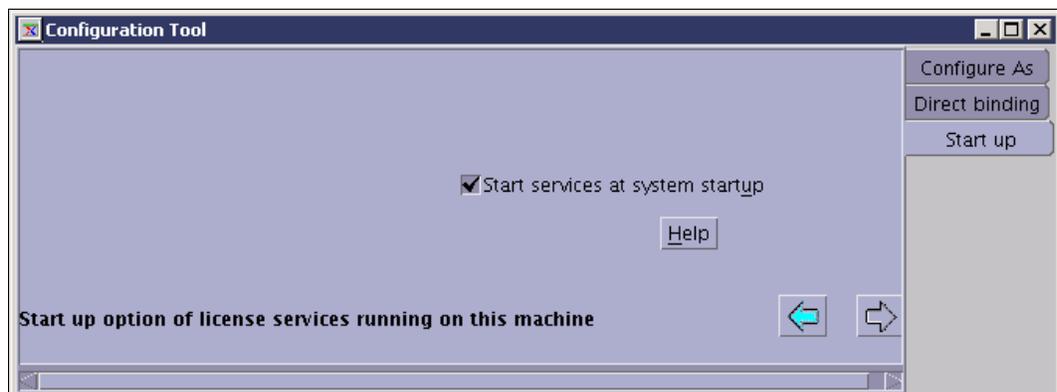


Figure 3-12 License Use Management: Start services at system startup option selected

20. Close the configuration Tool and save the changes to complete the License Use Management configuration.

21. Start License Use Management runtime by using the following command:

```
i4cfg -start
```

Example 3-14 shows that License Use Management was successfully started at our ITSO installation and is ready to enroll and distribute the WebSphere Voice Response license.

*Example 3-14 License Use Management successfully started*

```
# i4cfg -start
i4cfg Version 4.6.7 AIX -- LUM Configuration Tool
(c) Copyright 1995-2003, IBM Corporation, All Rights Reserved
US Government Users Restricted Rights - Use, duplication or disclosure
restricted by GSA ADP Schedule Contract with IBM Corp.

0513-059 The i4lmd Subsystem has been started. Subsystem PID is 20118.
0513-059 The i4gdb Subsystem has been started. Subsystem PID is 67818.

'Start Services' has completed successfully
```

22. To enroll the number of WebSphere Voice Response licenses that you have purchased, complete these steps:

- a. Type the **i4b1t** command to start the Basic License Tool.
- b. Select **Products** → **Enroll** → **Single Product**.
- c. Select **Import**, to import the WebSphere Voice Response for AIX V4.2 dtaixchan.lic license file from the /usr/lpp/dirTalk/db/license directory.
- d. The Enroll Product window is now updated with the license details.
- e. From the Enroll License window, enter the relevant administrator details, the number of licenses and click **OK**. We used 24 licenses.
- f. After the licenses are enrolled, you see the Basic License Tool window (Figure 3-13). Right-click the **WebSphere Voice Response for AIX V4.2 license** and select **Distribute license**.

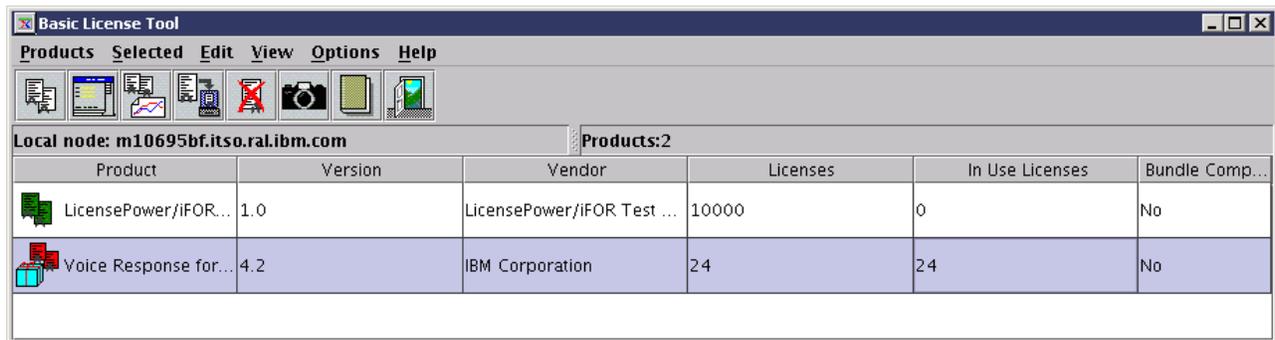


Figure 3-13 Basic License Tool window for licenses enrollment and distribution

- g. In the Distribute Licenses window, you see all the license servers that are available. Right-click **WebSphere Voice Response for AIX V4.2 license server** and click **Selected** → **Set number of licenses**.
- h. Enter the number of licenses to which you are entitled and click **OK**.

- i. In the Distribute Licenses window, click **Licenses** → **Distribute**. Then click **OK** to distribute the licenses. In our installation at the ITSO laboratory, we distributed 24 licenses.

This completes the license enrollment and distribution. You are now ready to start WebSphere Voice Response.

23. Before you start WebSphere Voice Response, check to see if some of its environmental parameters are set correctly. Type the following command to confirm that it is set to /usr/lpp/dirTalk:

```
echo $VAE
```

The \$DISPLAY variable is set according to the environment you are in. We set the \$DISPLAY variable to :0.0 to indicate that we are exporting to a local display.

If you have newly installed WebSphere Voice Response adapters, such as DTTA, DTEA, or DTXA, register dtuser as the owner.

If you installed any WebSphere Voice Response software since the last time you ran the **vae.setuser** command, you have to run the command again.

24. Log in as dtuser. You see the WebSphere Voice Response login menu (Example 3-15). Type 1 to start WebSphere Voice Response immediately.

*Example 3-15 WebSphere Voice Response 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) > 1
```

---

WebSphere Voice Response performs some processing. If the display is correctly set, the WebSphere Voice Response Status window opens. The host name of your machine also appears on the title bar. This status window generally remains open and displays WebSphere Voice Response startup messages and any messages related to the status of system. It is a DTstatus.out file in the /home/dirTalk/current\_dir/oamlog directory and is displayed by using the **tail** command. Every time WebSphere Voice Response restarts, the existing DTstatus.out file is saved into another file with the same name but with the date and time concatenated.

Example 3-16 shows the startup messages saved in the DTstatus.out file, which are also displayed in the WebSphere Voice Response Status window. You are asked to accept the terms of the license agreement if this is the first time you are starting WebSphere Voice Response.

*Example 3-16 DTstatus.out startup messages*

---

```
NM Welcome to WebSphere Voice Response from Node Manager (NM)
NM Starting initialization procedure
Checking fileset consistency
Checking import consistency
ssistatus: Querying configuration ..
ssistatus: m10695bf is a Standalone system
fscheck 190: Filesystem Check ok
Running some tests on DTXA 0 before we use it. (Takes approximately 1 minute)...
Tests on DTXA 0 complete.
dtdd Defined
dtdd Available
dtxa0 Defined
dtxa0 Available
dtxa0 Enabled
```

```
dtpack0_0 Defined
dtpack0_0 Available
dtpack0_1 Defined
dtpack0_1 Available
dtpack0_2 Defined
dtpack0_2 Available
dtpack0_3 Defined
dtpack0_3 Available
NM Checking installed subsystems and devices
NM: Running DTEA2 configuration
NM Task list read from /usr/lpp/dirTalk/db/sys_dir/tasklist.data
NM Loading program: DBIM
DBIM: Initialisation complete
NM Loading program: LRS
NM Loading program: OAM
NM Loading program: DBHEALTH
NM Loading program: SDI
NM Loading program: VAGIO
NM Loading program: STATEM
NM Loading program: PROMPTM
NM Loading program: CACHEM
NM Loading program: STPDSEVER
NM Loading program: VAGSEVER
NM Loading program: SGAMSEVER
NM Loading program: UPSERVER
NM Loading program: UDSEVER
NM Loading program: CHPM
NM Loading program: DTBEMGR
NM Loading program: CHP
NM Loading program: SCRSEVER
NM Loading program: SMSERVER
SCRSEVER: Initialization complete
NM Loading program: XREFSEVER
XREFSEVER: Initialization complete
NM Loading program: SLDAEMON
NM Loading program: CA_CNTL
NM Loading program: TSL0T
STPDSEVER: Initialization complete
NM Loading program: MWISEVER
NM Resource allocation complete
MWISEVER: Initialization complete
VAGSEVER: Initialization complete
dtpack0_0 Available
NM System initialization complete
UPSEVER: Initialization complete; pid=44642
VAGIO: Root process on display 9.42.170.139:0.0 is DT

VAGIO: Root process on display :0.0 is DT
ng
start_vae: WebSphere Voice Response starting
start_vae: WebSphere Voice Response starting
start_vae: WebSphere Voice Response starting
```

---

25. In the WebSphere Voice Response logon window (Figure 3-14) that opens, login as admin. The default password is admin. We recommend that you change the password immediately after logging in.



Figure 3-14 WebSphere Voice Response logon window

26. The WebSphere Voice Response Welcome window (Figure 3-15) opens, indicating that WebSphere Voice Response is ready for configuration. From the Welcome window, for this scenario, select **Configuration** → **Pack Configuration** → **Change**.



Figure 3-15 Successfully login to WebSphere Voice Response

27. For Region/Country, select your region or country. We selected **US/Canada**.

We also configured the WebSphere Voice Response trunk to perform the functions of ISDN\_5E11-5E15 National 2 user side protocol. The Trunk Parameters that we selected are shown in Figure 3-16. The remaining trunk configurations are shown in Figure 3-17. This configuration will interoperate with the Gordon Kapes System 930 Telephony Simulator.

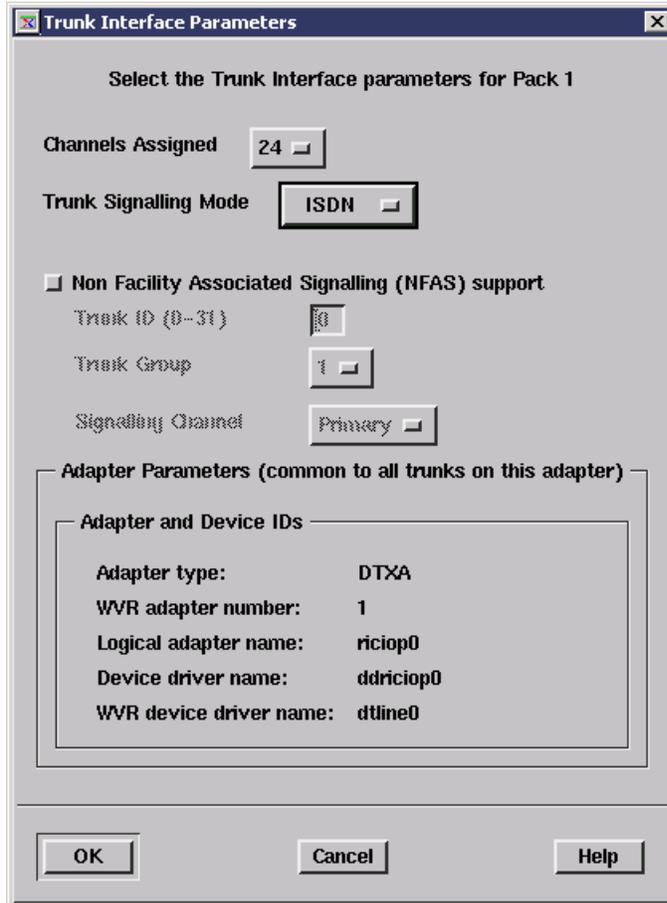


Figure 3-16 Trunk parameter selections

Since we do not want to receive calls immediately when WebSphere Voice Response is started, we set the Operating Status to **Available**. This blocks calls from reaching the WebSphere Voice Response and provides ample time for other applications in various systems to start. When all other systems were up and running, we placed the trunk in-service and saved the configuration. See Figure 3-17.

**Note:** This configuration is suitable for our test environment in the ITSO laboratory. Your business environment may require you to configure the WebSphere Voice Response differently.

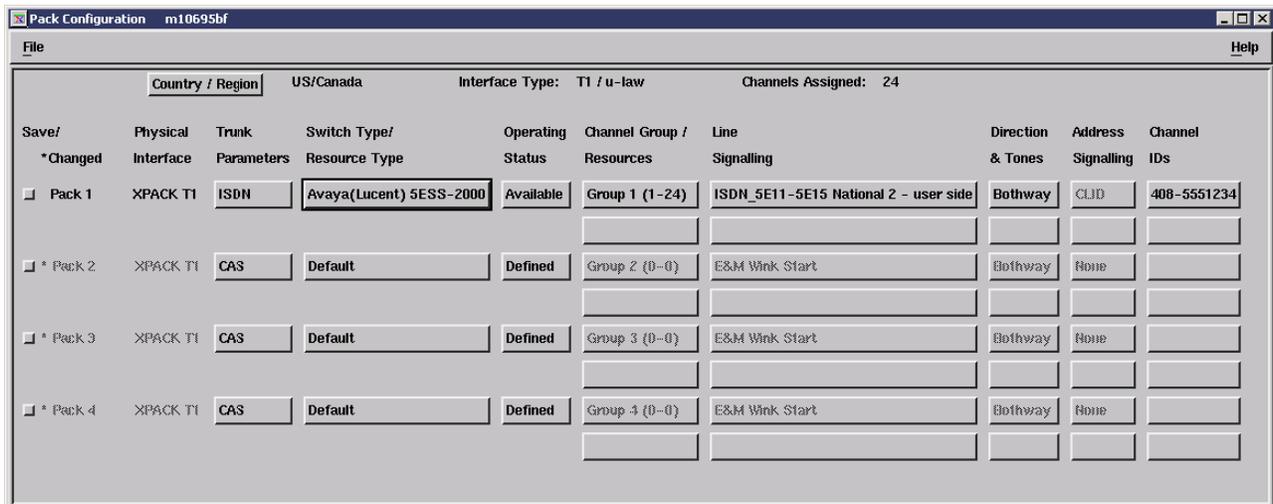


Figure 3-17 Pack configuration at ITSO laboratory

28. Because the ISDN protocol is used, we import the ISDN file set to establish a working ISDN T1 connection with the Gordon Kapes System 930 Telephony Simulator. With a working ISDN link, we eliminate ISDN link errors and alarms that are reported by the WebSphere Voice Response system and the telephony simulator.

In the Welcome window, click **Applications** → **Applications**.

29. In the Applications window (Figure 3-18), click **Applications** → **Import** → **Replace** → **File**.

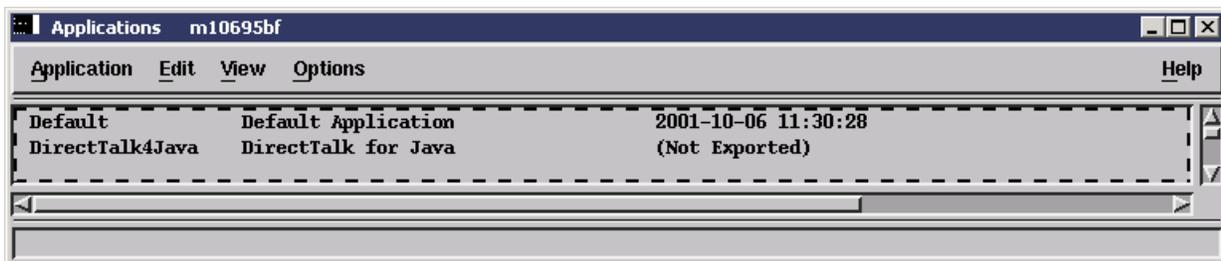


Figure 3-18 WebSphere Voice Response Applications window

30. In the Import File window, for the Search String field, type the path name `/usr/lpp/dirTalk/sw/isdn/*` and click **Search**.

The window displays the list of ISDN optional file sets as shown in Figure 3-19. We selected the **sigproc\_t1\_nat** file set and clicked **OK**.

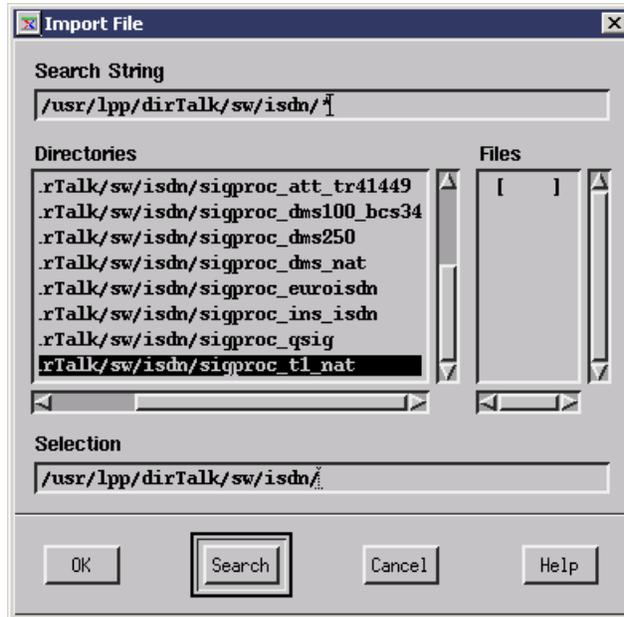


Figure 3-19 WebSphere Voice Response Import File window

31. The system imports the file and opens the Import Report window to show the status of the import. We saved the import report to a file. Example 3-17 shows the contents of that file.

*Example 3-17 ISDN sigproc\_t1\_nat file set import report*

---

```

Import Start: 2005/05/31 19:20:13
Importing Application Package from device
/usr/lpp/dirTalk/sw/isdn/sigproc_t1_nat/sigproc_t1_nat.imp
x ./db2export
x ./db2export/.version, 3498 bytes, 7 media blocks.
x ./db2export/.tableinfo, 2 bytes, 1 media blocks.
x ./db2export
x ./db2export/.version, 3498 bytes, 7 media blocks.
x ./db2export/.tableinfo, 2 bytes, 1 media blocks.
x ./db2export/appl, 6324 bytes, 13 media blocks.
x ./db2export/.appl_catalog, 6575 bytes, 13 media blocks.
x ./db2export/.stbl.001, 0 bytes, 0 media blocks.
x ./db2export/.pdir.001, 0 bytes, 0 media blocks.
x ./db2export/.prompt.001, 0 bytes, 0 media blocks.
x ./db2export/.voicetable.001, 0 bytes, 0 media blocks.
x ./db2export/.scr_screen_file.001, 0 bytes, 0 media blocks.
x ./db2export/.scr_func_file.001, 0 bytes, 0 media blocks.
x ./db2export/ca_info, 19574 bytes, 39 media blocks.
x ./db2export/ca_section_tbl, 6641 bytes, 13 media blocks.
x ./db2export/.ca_func_file.001, 0 bytes, 0 media blocks.
x ./db2export/subclass, 16627 bytes, 33 media blocks.
x ../current_dir/ca/T1_NATIONAL_ISDN_dir
x ../current_dir/ca/T1_NATIONAL_ISDN_dir/T1_NATIONAL_ISDN, 21155 bytes, 42 media blocks.
Imported      1 Rows into table appl
Imported      1 Rows into table ca_info
Imported      5 Rows into table ca_section_tbl

```

```
Imported      0 Rows into table subclass
Importing the T1_NATIONAL_ISDN custom server.
Installed Custom Server T1_NATIONAL_ISDN
Import End:   2005/05/31 19:20:15 RC=0
```

---

32. Because a Custom Server was installed, we need to make sure that it automatically starts.

- a. In the Welcome window, click **Operations** → **Custom Server Manager**.
- b. The T1\_NATIONAL\_ISDN custom server is displayed. In the Run Status column, click **start**. In the IPL Status column, click **Auto-Start On**. This automatically starts the ISDN custom server every time WebSphere Voice Response restarts.

For the changes to the configuration to be effective, restart WebSphere Voice Response. Note that not *every* change to the WebSphere Voice Response configuration requires a restart. The WebSphere Voice Response is designed to deliver high availability service with minimum disruption. Some parameters can be changed and activated while the system is still running.

33. To shut down WebSphere Voice Response, follow these steps:

- a. In the Welcome window, click **Operations** → **Immediate** to shut down WebSphere Voice Response immediately.
- b. Wait until you see the message Node Manager terminating on the WebSphere Voice Response Status window before you try to restart WebSphere Voice Response. This allows all the WebSphere Voice Response processes to terminate properly.
- c. Press Ctrl+C to close the WebSphere Voice Response Status window.

Another way to shut down WebSphere Voice Response is to log in as dtuser and type the command:

```
DT_shutdown
```

The shut down is complete when the AIX prompt is returned. Both the WebSphere Voice Response Welcome window and WebSphere Voice Response Status window close.

34. If you are already logged in as dtuser, start WebSphere Voice Response with the following command:

```
vaeinit
```

35. To enable the trunk, from the Welcome window, follow these steps:
- Click **Operations** → **System Monitor** to display the System monitor.
  - From the Trunk Status column, click **Available** → **Enable**.
  - In the Confirm Request window, click **OK** to enable the XPACK.

Figure 3-20 shows the windows for enabling the trunk.

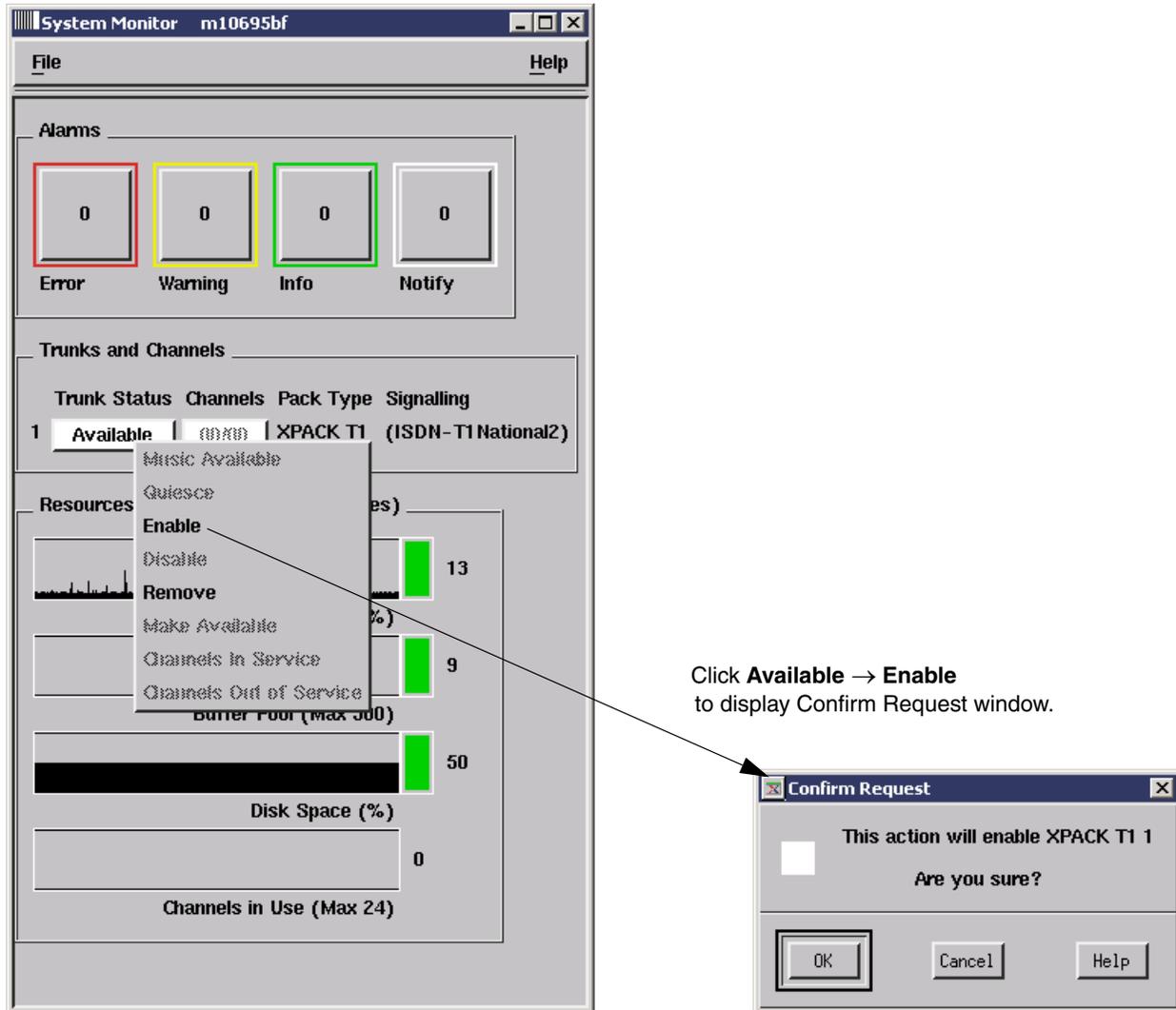


Figure 3-20 Enabling the trunk

- After some processing, the trunk is enabled and its status changes to *In Service*. From the Trunk Status column, click **In Service**.
- From the menu that appears, select **Channels In Service** to activate all the channels in the trunk.

Figure 3-21 shows all the channels are *In Service* except channel 24 which is reserved for ISDN signalling. This confirms that the DTXA card is communicating with the Gordon Kapes System 930 Telephony Simulator. Your installation may use a different trunk protocol which may not reserve channel 24 for signaling.

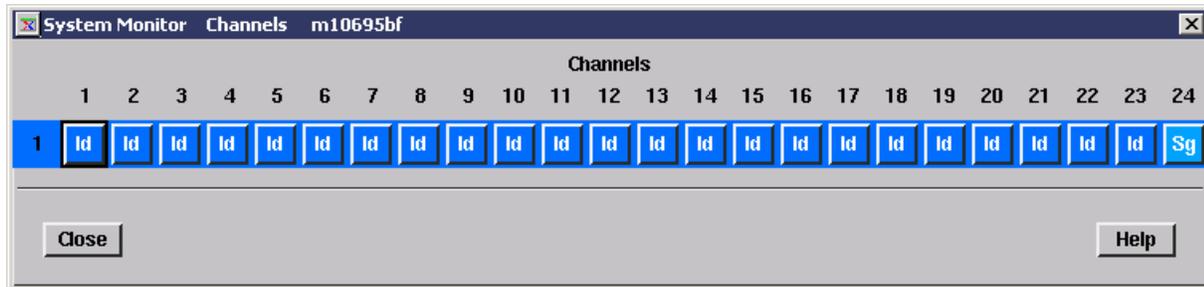


Figure 3-21 All 23 channels in the T1 trunk are *In Service idle.status*, with channel 24 reserved for ISDN signaling

36. Make a test call to the system. In our scenario, we heard the message: “Welcome to the IBM WebSphere Voice Response for AIX.” This proved that our installation was working.

37. Import the Java and VoiceXML applications.

Log in as dtuser. If we want to install additional language support to run Java applications other than U.S. English, use the `dtjn1sin` command from the `/var/dirTalk/DTBE/nls/mappers` directory, which contains the language zip files.

Example 3-18 shows a typical successful language installation.

*Example 3-18 Successful installation of Euro Italian language*

```
$ dtjn1sin it_IT_EURO.zip
2005.06.08 13:39:38 I DTJ3051 Successfully imported (category=System,name=EUR_ma
jorcur,locale=it_IT) at host LocalHost at node Node1.
2005.06.08 13:39:38 I DTJ3051 Successfully imported (category=System,name=EUR_ma
jorcurs,locale=it_IT) at host LocalHost at node Node1.
2005.06.08 13:39:38 I DTJ3051 Successfully imported (category=System,name=EUR_mi
norcur,locale=it_IT) at host LocalHost at node Node1.
2005.06.08 13:39:38 I DTJ3051 Successfully imported (category=System,name=EUR_mi
norcurs,locale=it_IT) at host LocalHost at node Node1.
vsegs/impexp.cfv -> /var/dirTalk/DTBE/nls/it_IT_EURO
The language pack has installed successfully.
```

In our installation for the ITS0, we installed the complete language support file sets.

38. To support speech technology, install the required plug-ins, which are stored in the `/var/dirTalk/DTBE/plugins` directory. The `dtjmrcp.zip` plug-in supports speech technology in WebSphere Voice Server V5.1.x. The `dtjvv.zip` plug-in supports the IBM WebSphere Voice Server V4.2 speech technology.

Because WebSphere Voice Server V5.1.x support is required in our installation, we installed `dtjmrcp.zip` plug-in using the following commands:

```
cd /var/dirTalk/DTBE/plugins
dtjplgin dtjmrcp.zip
```

We ensured that WebSphere Voice Response for AIX V4.2 was running but not the Java and VoiceXML node.

39. After the plug-in successfully installs, in the WebSphere Voice Response Welcome window, click **Operations** → **Custom Server Manager**.

40. In the Custom Server Manager window (Figure 3-22), start the custom servers and set them to Auto-Start.

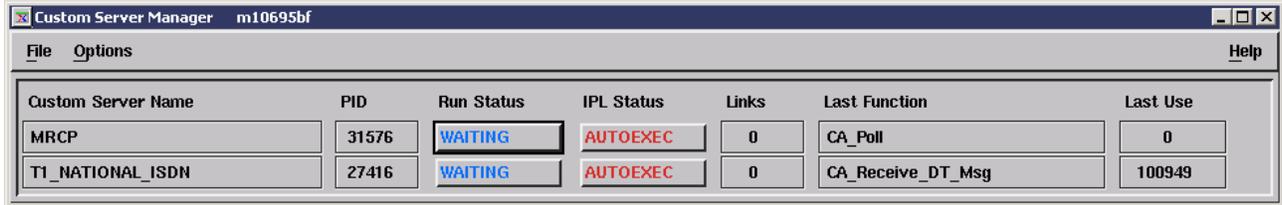


Figure 3-22 Custom Server Manager showing installed custom servers

41. To verify that the correct set of Custom Server User Functions is installed, follow these steps:

- a. In the Welcome window, select **Applications** → **Custom Servers**.
- b. In the Custom Servers window (Figure 3-23), under Custom Servers, select **MRCP** to view the User Functions on the right side.

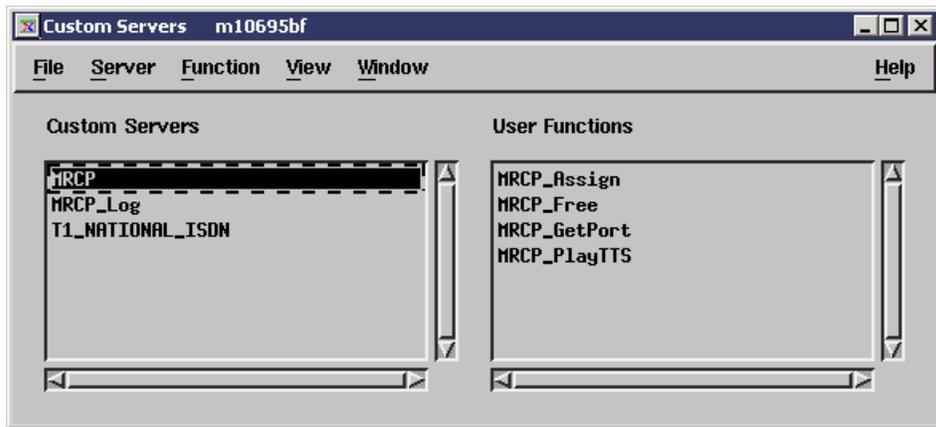


Figure 3-23 MRCP Custom Server User Functions

- c. Click **MRCP\_Log** and you see the respective User Functions as shown in Figure 3-24.

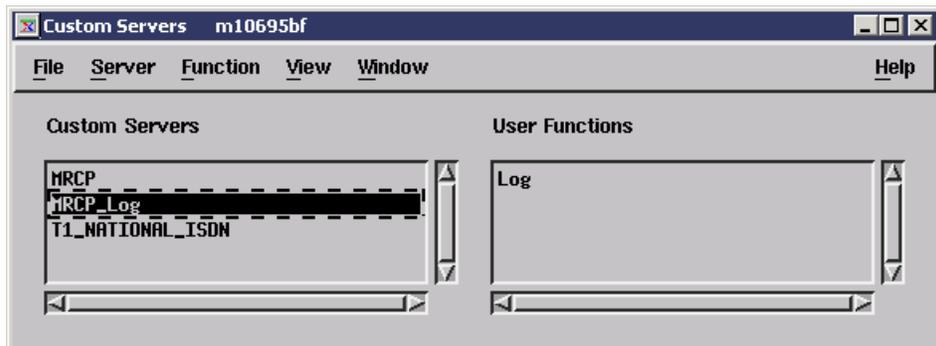


Figure 3-24 MRCP\_Log Custom Server User Functions

42. To check that the State Tables associated with the plug-in are successfully installed, complete these steps:
- In the Welcome window, click **Applications** → **Applications**.
  - In the Applications window (Figure 3-25), double-click the **MRCP** application.

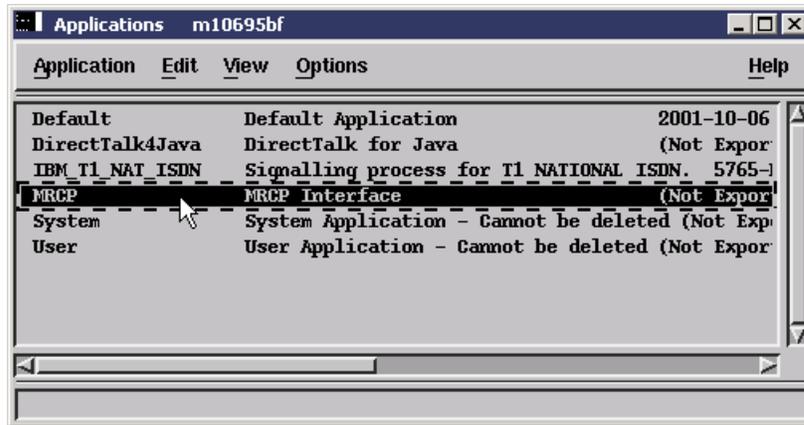


Figure 3-25 MRCP Interface

- In the Application (MRCP) window (Figure 3-26), click the **State Tables** icon to view the associated State Tables on the right side of the window. You can also click **Custom Servers** to display the list of Custom Servers associated with the MRCP application.



Figure 3-26 State Tables associated with MRCP Interface

43. Verify the installation by logging in as dtuser and running the following supplied script from the /var/dirTalk/DTBE/native/aix directory:

```
verify_install
```

Follow the instructions, like those shown in Example 3-19, and dial into the system. When the call was answered, we heard the recorded message: "Welcome, WebSphere Voice Response is now installed." Then the script completed.

*Example 3-19 Dialog of verify\_install script*

---

```
$ verify_install
Installation verification application is initializing without speech technology.
Please wait.
Waiting for call...
Installation verification application is ready.
Please dial into WebSphere Voice Response and wait for a few seconds.
Installation verification complete.
$ pwd
/var/dirTalk/DTBE/native/aix
```

---

### 3.7.5 Testing with WebSphere Voice Server V5.1.x

To test the Java and VoiceXML environments, you must start the WebSphere Voice Response Java host manager as well as the nodes and applications. The characteristics of the nodes and applications are defined in the default.cff file.

You can find a sample default.cff file in the /var/dirTalk/DTBE/native/aix directory. Most likely you have to modify it to suit your environment.

### 3.7.6 Considerations when deploying speech technology

Pay attention to the speech quality when you deploy speech recognition technology in a voice network. In an uncompressed PCM digital transmission, the speech signal is 64 Kbps mu-law or A-law encoded. This signal is generally good barring any interferences.

Speech compression in a TDM or IP environment is commonly deployed to conserve transmission bandwidth. This inevitably introduces some distortion to the original speech signal, which in turn affects the accuracy of the speech recognition technology. Some voice networks have multi-stage compression and decompression. This certainly will impact the speech quality. When designing speech technology in the voice network, you must understand the network topology, the users, and the type of telephone equipment they are using to access your service.

To improve the accuracy of speech recognition technology, we recommend that you allocate some time for the tuning process. Spend some time to fine-tune the grammar in the speech technology, which will improve the accuracy of your speech recognition engine.



## Testing our installation

This chapter describes the interoperability testing between WebSphere Voice Response and WebSphere Voice Server. It shows the test equipment that we used for testing some simple VoiceXtensible Markup Language (VoiceXML) applications. It also explains how to start and stop logging.

## 4.1 Equipment used in ITSO testing

Figure 4-1 shows the setup that we used in the International Technical Support Organization (ITSO) laboratory. WebSphere Voice Response for AIX V4.2 and IBM WebSphere Voice Server V5.1.2 were installed and configured to the test conditions in the ITSO. The VoiceXML application used for the tests is shown in Appendix A, “Sample code” on page 61.

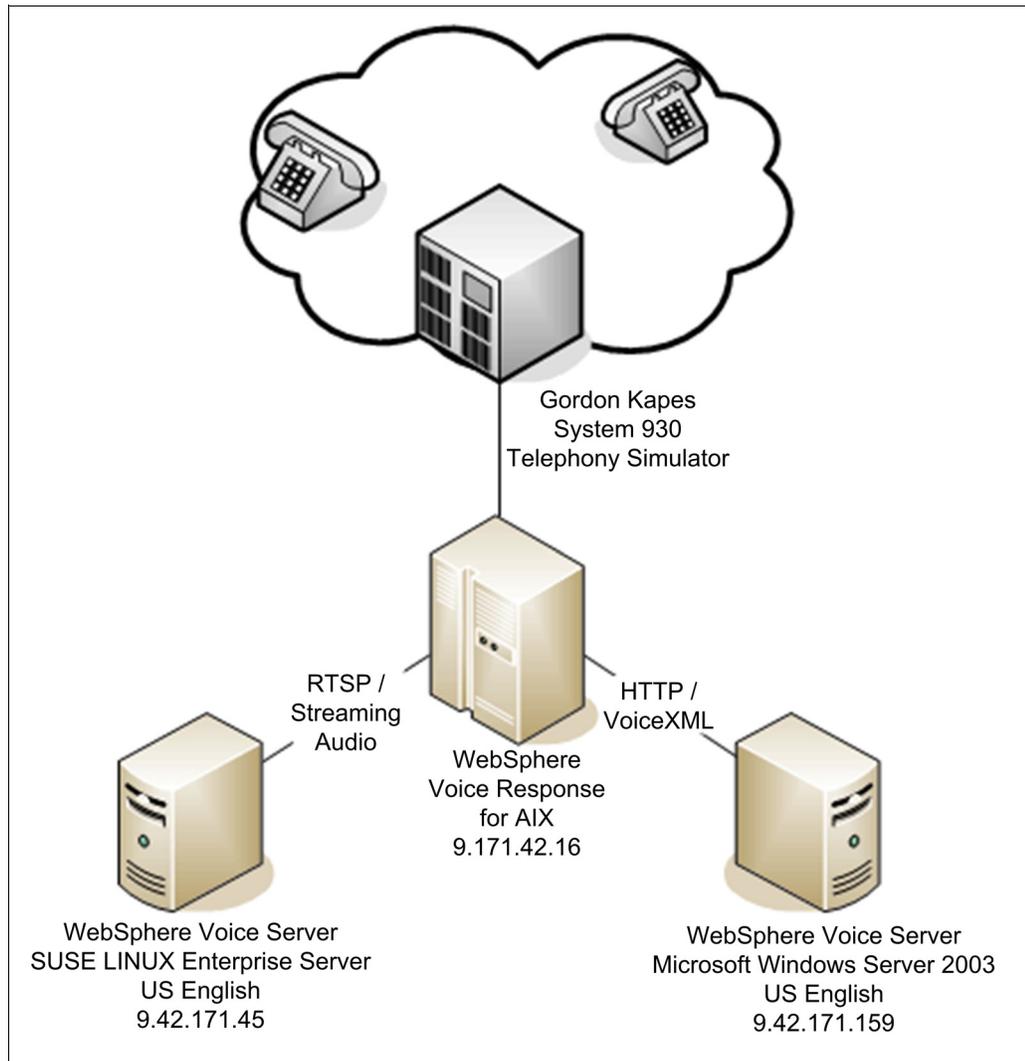


Figure 4-1 Configuration used for the tests done in the ITSO lab

## 4.2 Test scenarios

We performed tests to confirm that the text-to-speech (TTS) and automatic speech recognition (ASR) services were functioning. The services were provided by the IBM WebSphere Voice Server V5.1.2 that was installed on SUSE Linux Enterprise Server with IP address 9.42.171.45. We performed the test using with VoiceXML documents. These documents were fetched from the WebSphere Application Server that was running on Microsoft Windows 2003 Server with IP address 9.42.171.159 and locally from WebSphere Voice Response.

The default.cff file in the /var/dirTalk/DTBE/native/aix directory was modified to suit the test conditions. The default.cff listing shown in Example 4-1 is used to fetch the planetworldXML.vxml document from the WebSphere Application Server when a call is answered. The VoiceXML document then requests the TTS and ASR services from the IBM WebSphere Voice Server V5.1.2 installed on the SUSE Linux Enterprise Server. The TTSService and RecoService key words define the location and characteristics of the speech technology resources.

*Example 4-1 default.cff file*

---

```

TTSService=Tts_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPTTS
  InitSessionString=URI=rtsp://9.42.171.45/media/synthesizer
  TTSType=Ttsen_US
;

RecoService=Reco_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPreco
  InitSessionString=URI=rtsp://9.42.171.45/media/recognizer
  RecoType=Recoen_US
;

AppName=CCXMLVoiceXMLBrowser
  Enabled=yes
  AppClass=com.ibm.wvr.vxml2.DTVoicelet2
  Parameter=DynamicURI,yes
;

AppName=RecoTTS
  Enabled=yes
  AppClass=com.ibm.wvr.vxml2.DTVoicelet2
  # Parameter=URI, file:///home/dtuser/test/RecoTTSenUS.vxml
Parameter=URI, http://9.42.171.159/planets/planetworldXML.vxml
;

GroupName=group1
  Application=CCXMLVoiceXMLBrowser
;

GroupName=RecoTTSGroup
  Enabled=yes
  Application=RecoTTS
  Application=RecoTTS
;

# This is the definition of a Voice Response node:
NodeName=Node1
  Enabled=yes
  NodeDefLocale=en_US
  VRNode=yes
  NodeDefAppName=RecoTTS
  RecoService=Reco_US
  RecoDefinition=en_US,Recoen_US
  TtsService=Tts_US
  TtsDefinition=en_US,Ttsen_US
  NumToApp=*,RecoTTS
  JavaCommand=java -Xmx128m
;

# This is the definition of an application node:

```

```

NodeName=AppNode1
  Enabled=yes
  VRNode=no
  NodeDefHost=localhost
  NodeDefVRNode=Node1
  Group=RecoTTSGroup
  #Group=group1
  JavaCommand=java -Xmx128m
;

# This is the definition of the local host:
HostName=localhost
  Enabled=yes
  IPName=localhost
  Node=Node1
  Node=AppNode1
;

```

---

To change the source of the VoiceXML document to be fetched, we modified the definition of the secondary keyword, `Parameter`, belonging to the key word `AppName` as highlighted in Example 4-1. For example, to fetch the VoiceXML document from the WebSphere Voice Response, we used this statement:

```
Parameter=URI, file:///home/dtuser/test/RecoTTSenUS.vxml
```

To update the changes in the `default.cff` file to the configuration database, we issued the **dtjconf** command. Then we stopped the node manager by using the **dtjstop** command. Finally, we restarted the node manager by entering the **dtjstart** command. Example 4-2 shows the sequence of commands and responses when updating the configuration database with a new copy of `default.cff` file.

*Example 4-2 Updating the configuration database and restarting the nodes*

---

```

$ dtjconf
Running the ConfigManager to import the default configuration ...
Configuration default has been successfully added from default.cff to config.cfd

$ dtjstop
Running the PlexManager to stop the LocalHost ...
2005.06.09 13:40:30 I DTJ3031 Stopping node Node1 at host localhost.
2005.06.09 13:40:30 I DTJ3011 Node Node1 at host localhost has stopped.
2005.06.09 13:40:30 I DTJ3031 Stopping node AppNode1 at host localhost.
2005.06.09 13:40:30 I DTJ3011 Node AppNode1 at host localhost has stopped.

$ dtjstart
Running the PlexManager to start the LocalHost ...
2005.06.09 13:40:36 I DTJ3015 Starting voice response node Node1 at host localhost.
2005.06.09 13:40:39 I DTJ3016 Voice response node Node1 at host localhost has started.
2005.06.09 13:40:40 I DTJ3030 Starting applications for node Node1 at host localhost.
2005.06.09 13:40:40 I DTJ3028 Starting node AppNode1 at host localhost.2005.06.09 13:40:42
I DTJ3010 Node AppNode1 at host localhost has started.
2005.06.09 13:40:42 I DTJ3030 Starting applications for node AppNode1 at host localhost.
2005.06.09 13:40:42 I DTJ3019 Application RecoTTS was started on node AppNode1 at host
localhost.
2005.06.09 13:40:42 I DTJ3019 Application RecoTTS was started on node AppNode1 at host
localhost.
$

```

---

To confirm that the correct VoiceXML application started correctly, we made test calls to the WebSphere Voice Response.

To assist with our testing, we turned on the trace facilities in the WebSphere Voice Response and IBM WebSphere Voice Server V5.1.2. In a full production environment, we recommend that you do not turn on traces.

## 4.2.1 Testing CCXML

WebSphere Voice Response for AIX V4.2 supports Call Control Extensible Markup Language (CCXML). We used the following files and documents to test the CCXML installation:

- ▶ default.cff
- ▶ /home/dtuser/test/directcall.ccxml
- ▶ /home/dtuser/test/RecoTTSenUS.vxml
- ▶ /home/dtuser/test/outgoingcall.vxml

The listing of these files is shown in Appendix A, “Sample code” on page 61.

In the default.cff file, the secondary key word InitialURI under the CCXService key word points to the directcall.ccxml document. When a call arrives, the directcall.xml document processes the call according to the number dialed.

If the called number is 8182838, the document answers the call and then hangs up. It then generates any event that automatically make calls. If the called number is 8182839, the document sends an event to stop to disconnect the calls that are generated earlier. If the called number is neither of those two numbers, it fetches the RecoTTSenUS.vxml document.

## 4.2.2 WebSphere Voice Response for AIX V4.2 logs and traces

WebSphere Voice Response stores the log and trace binary messages in the log and trace files in the /var/dirTalk/DTBE/dtj\_log directory. The name of the log files is log.n.log and the name of the trace files is wvrtrace.n.trc where *n* has the value 1 to 10. These files are 5 MB each by default. Your file system must be configured to allocate enough space for these files.

You may format the log file by using the **dtjfllog** command. For example, to format the binary message file, log.8.log, to a text file, log8.txt, use the following command:

```
dtjfllog -a -o log8.txt log.8.log
```

To format the binary trace file, wvrtrace.3.trc, to a readable format and save it in the file, wvrtrace3.txt, use the following command:

```
dtjtrcfmt wvrtrace.3.trc > wvrtrace3.txt
```

In our environment, we have the WebSphere Voice Response speech technology log dynamically updated on the screen. We used the following command to do that:

```
tail -f log.1.log | dtjfllog
```

The file log.1.log may not be the active log in your installation at the time when the command is issued. In that case, go to the /var/dirTalk/DTBE/dtj\_logs directory and check which log.n.log file is most recently written into.

Example 4-3 shows a truncated log in response to the previous `dtjstop` and `dtjstart` commands that were issued.

*Example 4-3 Truncated log showing the stop and start of Node1*

---

```
2005-06-09 13:59:34.000499 call 0:0.0.0.0 PID[0:3] (DTJ7002) INFO: VXML browser timed out
waiting for call - delaying 30000ms before cycling.
2005-06-09 13:59:36.000093 call 0:0.0.0.0 PID[0:2] (DTJ2006) INFO: Quiesce shutdown for
node AppNode1 at host LocalHost has started.
2005-06-09 13:59:39.000066 call 0:0.0.0.0 PID[0:12] (DTJ2008) INFO: Node Node1 at host
LocalHost has shutdown.
2005-06-09 13:59:39.000130 call 0:0.0.0.0 PID[0:30] (DTJ2008) INFO: Node AppNode1 at host
LocalHost has shutdown.
2005-06-09 13:59:52.000319 call 0:0.0.0.0 PID[0:1] (DTJ2002) INFO: Creating a new
ApplicationManagerImpl for node Node1 at host LocalHost.
2005-06-09 13:59:52.000818 call 0:0.0.0.0 PID[0:1] (DTJ2004) INFO: ApplicationManagerImpl
for node Node1 at host LocalHost successfully created.
2005-06-09 13:59:54.000075 call 0:0.0.0.0 PID[0:2] (DTJ7027) INFO: vxml2.DTAudioCache
Initialising
2005-06-09 13:59:54.000373 call 0:0.0.0.0 PID[0:2] (DTJ7028) INFO: vxml2.DTAudioCache
Initialisation complete - Loaded successfully
2005-06-09 13:59:56.000305 call 0:0.0.0.0 PID[0:1] (DTJ2002) INFO: Creating a new
ApplicationManagerImpl for node AppNode1 at host LocalHost.
2005-06-09 13:59:56.000814 call 0:0.0.0.0 PID[0:1] (DTJ2004) INFO: ApplicationManagerImpl
for node AppNode1 at host LocalHost successfully created.
```

---

## 4.3 WebSphere Voice Server trace functions

You can verify any difficulties with the audio between the WebSphere Voice Server and WebSphere Voice Response by issuing a debug on the *MediaConv* Mbean, which is the audio interpreter. When setting the *MediaConv* from the IBM WebSphere Voice Server tree to debug, all the audio received by the ASR and produced by the TTS is recorded to a file in the `$WAS_ROOT/logs/voice-server/CWV/timestamp` directory. A new directory with a time stamp is created each time the voice server is restarted. This produces a lot of data and should remain turned on.

Debugging RTSP traffic shows the setup and teardown of steaming audio transactions. The error in the RTSP traffic indicates whether a request does not pass authentication. If the RTSP negotiations are okay, then try the *MediaConv* debugging as described previously. Having both the RTSP and the *MediaConv* output checked as okay determine that an audio problem may lie in the telephony interface of WebSphere Voice Response. Figure 4-2 shows setting the trace attribute for *RTSPBridge* to *event + debug*.

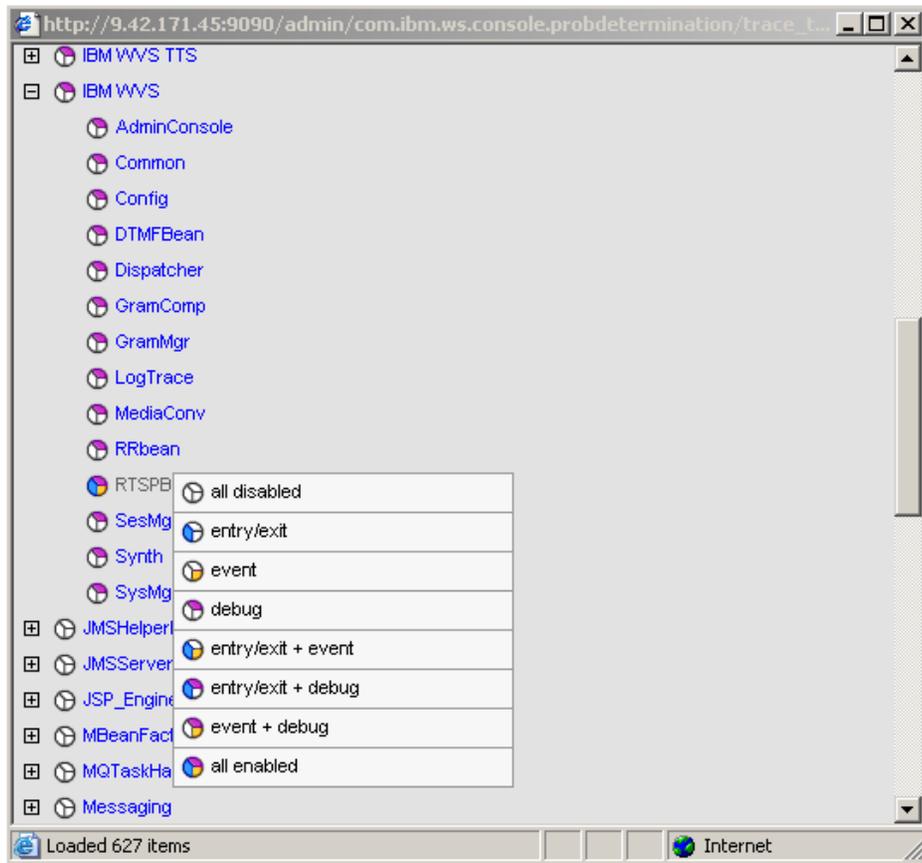


Figure 4-2 Trace attribute definition list

Example 4-4 shows a partial log extract from tracing RTSPBridge. In this example, a recognition request is validated and passes. It is followed by the URI it was called on.

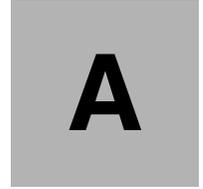
*Example 4-4 Partial log extract from tracing RTSPBridge*

---

```
[6/9/05 22:33:18:635 EDT] 4d89e32b RTSPBridge > RTSPRequest validate [sid=null][rid=-1] Entry
[6/9/05 22:33:18:635 EDT] 4d89e32b RTSPBridge < RTSPRequest validate [sid=null][rid=-1] Exit
                    success
[6/9/05 22:33:18:635 EDT] 4d89e32b RTSPBridge > RTSPSessionObj processRequest [sid=null][rid=-1] Entry
                    DESCRIBE rtsp://kpcrr06/media/recognizer RTSP/1.0
```

---





## Sample code

This appendix provides sample code that was used in testing and evaluating WebSphere Voice Response for AIX V4.2.

## Default.cff for testing TTS and ASR

Example A-1 shows the default.cff file that we used to test text to speech (TTS) and automatic speech recognition (ASR).

*Example: A-1 default.cff for speech technology testing*

---

```
TTSService=Tts_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPTTS
  InitSessionString=URI=rtsp://9.42.171.45/media/synthesizer
  TTSType=Ttsen_US
;

RecoService=Reco_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPreco
  InitSessionString=URI=rtsp://9.42.171.45/media/recognizer
  RecoType=Recoen_US
;

AppName=CCXMLVoiceXMLBrowser
  Enabled=yes
  AppClass=com.ibm.wvr.vxml2.DTVoicelet2
  Parameter=DynamicURI,yes
;

AppName=RecoTTS
  Enabled=yes
  AppClass=com.ibm.wvr.vxml2.DTVoicelet2
  Parameter=URI, http://9.42.171.159/planets/planetworldXML.vxml
;

GroupName=group1
  Application=CCXMLVoiceXMLBrowser
;

GroupName=RecoTTSGroup
  Enabled=yes
  Application=RecoTTS
  Application=RecoTTS
;

# This is the definition of a Voice Response node:
NodeName=Node1
  Enabled=yes
  NodeDefLocale=en_US
  VRNode=yes
  NodeDefAppName=RecoTTS
  RecoService=Reco_US
  RecoDefinition=en_US,Recoen_US
  TtsService=Tts_US
  TtsDefinition=en_US,Ttsen_US
  NumToApp=*,RecoTTS
  JavaCommand=java -Xmx128m
;

# This is the definition of an application node:
NodeName=AppNode1
  Enabled=yes
  VRNode=no
  NodeDefHost=localhost
```

```

NodeDefVRNode=Node1
Group=RecoTTSGroup
JavaCommand=java -Xmx128m
;

# This is the definition of the local host:
HostName=localhost
Enabled=yes
IPName=localhost
Node=Node1
Node=AppNode1
;

```

---

## Default.cff for testing CCXML

Example A-2 shows the default.cff file used to test Call Control Extensible Markup Language (CCXML).

*Example: A-2 default.cff file used to test CCXML*

---

```

#-----
# File Name      : default.cff
# Description    : default configuration file for IBM WebSphere Voice Response for AIX
#-----
# Required entry
#-----

TTSService=Tts
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPTTS
  InitSessionString=URI=rtsp://9.42.171.72/media/synthesizer
  TTSType=Ttsen_GB
;

TTSService=Tts_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPTTS
  InitSessionString=URI=rtsp://9.42.171.72/media/synthesizer
  TTSType=Ttsen_US
;

RecoService=Reco
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPreco
  InitSessionString=URI=rtsp://9.42.171.72/media/recognizer
  RecoType=Recoen_GB
;

RecoService=Reco_US
  PlugInClass=com.ibm.telephony.directtalk.mrcp.MRCPreco
  InitSessionString=URI=rtsp://9.42.171.72/media/recognizer
  RecoType=Recoen_US
;

#CCXML Service definition
CCXService=ccxml1
  Enabled=yes
  InitialURI=file:///home/dtuser/test/directcall.ccxml
  DefAppService=Node1
  CacheLimit=1M
  AppNodeThreads=0,24

```

```

VRNodeThreads=1,30
# Parameter=Key,Value
;
TelURLLocale=hursley
GlobalContext=+
LocalContext=01962
InternationalAccess=9
;

AppName=CCXMLVoiceXMLBrowser
Enabled=yes
AppClass=com.ibm.wvr.vxml2.DTVoicelet2
Parameter=DynamicURI,yes
;

AppName=dyn
Enabled=yes
AppClass=com.ibm.wvr.vxml2.DTVoicelet2
Parameter=DynamicURI,yes
;

GroupName=group1
Enabled=yes
Application=CCXMLVoiceXMLBrowser
Application=CCXMLVoiceXMLBrowser
;

GroupName=group2
Enabled=yes
Application=dyn
Application=dyn
Application=dyn
Application=dyn
;

GroupName=group3
Enabled=yes
Application=dyn
Application=dyn
Application=dyn
;

GroupName=group4
Enabled=yes
Application=dyn
Application=dyn
Application=dyn
;

# This is the definition of a Voice Response node:

NodeName=Node1
Enabled=yes
NodeDefLocale=en_GB
VRNode=yes
NodeDefAppName=ccxml1
CCXAppService=ccxml1
RecoService=Reco
RecoService=Reco_US
RecoDefinition=en_GB,Recoen_GB

```

```

RecoDefinition=en_US,Recoen_US
TtsService=Tts
TtsService=Tts_US
TtsDefinition=en_GB,Ttsen_GB
TtsDefinition=en_US,Ttsen_US
NumToApp=*,ccxm11
JavaCommand=java -Xmx128m
;

# This is the definition of an application node:

NodeName=AppNode1
  Enabled=yes
  VRNode=no
  NodeDefHost=LocalHost
  NodeDefVRNode=Node1
  Group=group1
;

NodeName=AppNode2
  Enabled=yes
  VRNode=no
  NodeDefHost=LocalHost
  NodeDefVRNode=Node1
  Group=group2
  JavaCommand=java -Xmx128m
;

NodeName=AppNode3
  Enabled=yes
  VRNode=no
  NodeDefHost=LocalHost
  NodeDefVRNode=Node1
  Group=group3
  JavaCommand=java -Xmx128m
;

NodeName=AppNode4
  Enabled=yes
  VRNode=no
  NodeDefHost=LocalHost
  NodeDefVRNode=Node1
  Group=group4
  JavaCommand=java -Xmx128m
;

#-----
# HostName entries
#-----

# This is the definition of the local host:

```

```

HostName=localhost
Enabled=yes
IPName=localhost
Node=Node1
Node=AppNode1
Node=AppNode2
Node=AppNode3
Node=AppNode4
;

```

---

## RecoTTSenUS.vxml

Example A-3 shows the VoiceXML used for testing TTS and ASR. It was also used in CCXML testing.

*Example: A-3 RecoTTSenUS.vxml document for TTS and speech recognition tests*

---

```

<?xml version="1.0" encoding="iso-8859-1"?>
<vxml version="2.0" xml:lang="en-US" xmlns="http://www.w3.org/2001/vxml">
  <!-- Voice bargein on TTS prompt -->
  <property name="bargein" value="false"/>
  <property name="sensitivity" value="0.2"/>
  <form>
    <block>
      <log>WK2.About to start testcase</log>
      About to start testcase. This prompt has no barge-in to set up echo
      cancellation.
    </block>
    <field name="userinput">
      <grammar root="words" version="1.0">
        <rule id="words" scope="public">
          <one-of>
            <item>Southampton</item>
            <item>Winchester</item>
            <item>Bristol</item>
            <item>London</item>
            <item>Manchester</item>
            <item>Hereford</item>
            <item>Oxford</item>
            <item>Guildford</item>
            <item>Durham</item>
          </one-of>
        </rule>
      </grammar>
      <prompt bargein="true" bargeintype="speech">
        This prompt is designed to test voice recognition using speech
        barge in over a text to speech prompt. Please interrupt this
        prompt at any time by saying Birmingham. This prompt will now
        count to five. 1, 2, 3, 4, 5.
      </prompt>
      <filled>
        <if cond="userinput == 'Birmingham'">
          <log>Testcase passed</log>
          Correctly recognised <value expr="userinput"/>.
          <exit/>
        <else/>
          <log>Wrong utterance - (<value expr="userinput"/>)</log>
          I thought you said <value expr="userinput"/>. I was expecting

```

```

        Birmingham.
    <exit/>
</if>
</filled>
<noinput>
    <log>No input.</log>
    No input received.
    <exit/>
</noinput>
<nomatch>
    <log>No match.</log>
    No match received.
    <exit/>
</nomatch>
</field>
</form>
</vxml>

```

---

## directcall.ccxml

Example A-4 shows how to trigger events or fetch another document based on the called number. If the called number is 8182838, it fetches the makecall.ccxml document. If it is 8182839, it sends a disconnect event to disconnect calls made earlier. Any other called number triggers the RecTTSenUS.vxml document.

*Example: A-4 dircetcall.ccxml document*

---

```

<?xml version="1.0" encoding="UTF-8"?>
<ccxml version="1.0">

    <var name="sessionid"/>
    <var name="secondsessionid"/>
    <eventprocessor>

        <transition event="ccxml.loaded" name="evt">
            <assign name="sessionid" expr="evt.sessionid"/>
            <log expr="'Direct call CCXML script started'"/>
        </transition>

        <transition event="connection.alerting" name="evt">
            <log expr="'Incoming call into application'"/>
            <accept connectionid="evt.connectionid"/>
        </transition>

        <transition event="connection.connected" name="evt">
            <log expr="'test..connection.connected received'"/>
            <if cond="evt.connection.local=='8182838'">
                <log expr="'Creating makecall ccxml document'"/>
                <createccxml next="makecall.ccxml" sessionid="secondsessionid"/>

                <disconnect connectionid="evt.connectionid"/>
                <elseif cond="evt.connection.local=='8182839'">
                    <log expr="'Send message to stop call generation'"/>
                    <send data="stopmakecall" target="secondsessionid"/>
                <disconnect connectionid="evt.connectionid"/>
            </if>
        </transition>

        <else/>
            <log expr="'Calling vxml document'"/>
        </else/>
    </eventprocessor>
</ccxml>

```

```

        <dialogstart src="'file:///home/dtuser/test/RecoTTSenUS.vxml'"/>
        </if>
    </transition>

    <transition event="dialog.exit" name="evt">
        <disconnect connectionid="evt.connectionid"/>
    </transition>

    <transition event="connection.disconnected" />
    <transition event="send.successful"/>
        <transition event="ccxml.created"/>

    <transition event="error.*" name="evt">
        <log label="'Warning'" expr="'error EVENT RECEIVED: '+evt.name"/>
        <log label="'Warning'" expr="'error: '+evt.reason"/>
    </transition>

    <transition event="*" name="evt">
        <log label="'Warning'" expr="'UNEXPECTED EVENT RECEIVED: '+evt.name"/>
    </transition>

</eventprocessor>
</ccxml>

```

---

## The makecall.ccxml document

Example A-5 is the CCXML document that is fetched when the called number 8182838 is detected by directcall.ccxml. It then generates a call every half second.

*Example: A-5 makecall.ccxml document*

---

```

<?xml version="1.0" encoding="UTF-8"?>
<ccxml version="1.0">
    <var name="sessionid"/>
    <var name="count" expr="0"/>
    <var name="running" expr="0"/>
    <eventprocessor>

        <transition event="ccxml.loaded" name="evt">
            <log expr="'MAKECALL: Call generation CCXML script started'"/>
            <assign name="sessionid" expr="evt.sessionid"/>
            <assign name="running" expr="1"/>
            <send data="'makecall'" target="evt.sessionid" namelist="sessionid"/>
        </transition>

        <transition event="makecall" name="evt">
            <log expr="'MAKECALL: makecall event received'"/>
            <if cond="count!=1000">
                <if cond="running==1">
                    <log expr="'MAKECALL: doing createcall'"/>
                    <createcall dest="'tel:57776662;phone-context=01962'"/>
                </if>
            </if>
        </transition>

        <transition event="stopmakecall" name="evt">
            <log expr="'MAKECALL: stopmakecall event received'"/>
            <assign name="running" expr="0"/>
        </transition>
    </eventprocessor>
</ccxml>

```

```
</transition>

<transition event="connection.progressing"/>
<transition event="send.successful"/>

<transition event="connection.connected" name="evt">
  <log expr="'MAKECALL: connection.connected event received'"/>
  <send data="'makecall'" target="sessionid" delay="'500ms'"/>
  <assign name="count" expr="count+1"/>
  <dialogstart src="'file:///home/dtuser/test/outgoingcall.vxml'"/>
</transition>

<transition event="dialog.started" />

<transition event="dialog.exit" name="evt">
<log expr="'MAKECALL: dialog.exit event received'"/>
  <disconnect connectionid="evt.connectionid"/>
</transition>

<transition event="connection.disconnected" >
  <log expr="'MAKECALL: connection.disconnected received'"/>
</transition>

<transition event="error.*" name="evt">
  <log label="'MAKECALL: Warning'" expr="'error EVENT RECEIVED: '+evt.name"/>
  <log label="'MAKECALL: Warning'" expr="'error: '+evt.reason"/>
</transition>

<transition event="*" name="evt">
  <log label="'MAKECALL: Warning'" expr="'UNEXPECTED EVENT RECEIVED: '+evt.name"/>
</transition>

</eventprocessor>
</ccxml>
```

---



# Glossary

**ANI** See *automatic number identification*.

**ASR** See *automatic speech recognition*.

**automatic number identification (ANI)** A service that tells the recipient of a telephone call, the telephone number of the person making the call.

**automatic speech recognition (ASR)** Speech recognition technologies that allow computers equipped with a source of sound input, such as a telephone handset or microphone, to interpret human speech.

**barge-in** A capability that allows callers to speak or enter their responses during the prompt.

**Call Control Extensible Markup Language (CCXML)**  
An XML standard designed to provide telephony support to VoiceXML. Where VoiceXML is designed to provide a voice user interface (VUI) to a voice browser, CCXML is designed to inform the voice browser how to handle the telephony control of the voice channel. The two languages are wholly separate and are not required by each other to be implemented.

**call flow** The “path” that a caller is directed through when interacting with a voice response system.

**CCXML** See *Call Control Extensible Markup Language*.

**cluster** A group of locally connected computers that work together as a unit.

**component** A pre-defined segment of a voice-driven application.

**Computer Telephony Integration (CTI)** The name given to the merger of traditional telecommunications (PBX) equipment with computers and computer applications. The use of caller ID to retrieve customer information automatically from a database is an example of a CTI application.

**confidence score** A value that indicates how certain the recognizer is that the result is correct. The confidence value can be used by the application to decide whether to proceed without asking for confirmation of the data, whether to ask for confirmation of the data, or whether to ask the caller to repeat the data.

**configuration** The arrangement of the software and hardware in a computer system or network.

**confirmation** Short dialog flow that repeats a recognition result back to the caller and asking the caller to confirm that it is correct.

**CTI** See *Computer Telephony Integration*.

**database** A collection of information stored in a computer in a systematic way, so that a computer program can consult it to answer questions.

**debug** The process of locating and correcting errors in computer programs.

**default** A particular setting or value for a variable that is assigned automatically by an operating system and remains in effect unless canceled or overridden by the operator.

**dialed number identification service (DNIS)** A telephone function that sends a dialed telephone number to an answering service.

**dialog** Interactions between the caller and the application.

**dialog box** A secondary window in which you are asked to fill in information.

**directory** A type of file used to group and organize other files or directories. Also called a *folder*.

**DNIS** See *dialed number identification service*.

**DTMF** See *dual-tone multifrequency tones*.

**dual-tone multifrequency (DTMF) tones** The system used by Touch-Tone™ telephones. DTMF assigns a specific frequency (made up of two separate tones) to each key so that it can easily be identified by a microprocessor. This is the technology behind Touch-Tone dialing.

**form** A document (printed or electronic) with spaces in which to write or enter data.

**graphical user interface (GUI)** A user interface based on graphics (icons, pictures, and menus) instead of text.

**GUI** See *graphical user interface*.

**hardware** The physical components of a computer system, such as keyboards, monitors, and media disk drives.

**Integrated Services Digital Network (ISDN)** An international standard for end-to-end digital transmission of voice, data and signaling.

**interactive voice response (IVR)** A telecommunications system that uses prerecorded voice messages to present options to a user.

**interface** The point of interaction or communication between a computer and any other entity, such as a printer or human operator.

**ISDN** See *Integrated Services Digital Network*.

**IVR** See *interactive voice response*.

**J2EE** See *Java 2 Platform, Enterprise Edition*.

**Java 2 Platform, Enterprise Edition (J2EE)** An environment for developing and deploying enterprise applications, defined by Sun™ Microsystems™ Inc. The J2EE platform consists of a set of services, application programming interfaces (APIs), and protocols that provide the functionality for developing multi-tiered, Web-based applications.

**menu** A GUI element which allows the user to select one among several, presumably related, choices.

**module** A separate and distinct unit of hardware or software that may be used as a component in a system.

**Natural Language Speech Recognition (NLSR)** An advanced type of speech recognition. NLSR can recognize particular words and phrases spoken by the user.

**NLSR** See *Natural Language Speech Recognition*.

**node** A device connected to a network, such as a computer or router.

**parameter** A value or object which is used to transfer information to or from subprograms. Input parameters can be thought of as being copied into the subprogram when it is called. Output parameters can be thought of as having their values copied back to the caller when the subprogram returns.

**PBX** See *private branch exchange*.

**platform** A platform refers to a framework on which applications may be run.

**port** A connection to a device that allows data or information to travel into and out of the device.

**predictive dialing** Automated dialing feature in which the systems predicts from pre-set algorithms when an agent or operator will become free and makes the call in advance.

**private branch exchange (PBX)** A private telephone switching system, usually located on a company's premises.

**prompt** A message played to a caller that gives the caller a choice of selections in a menu and asks for a response. Compare to *announcement*.

**proxy server** A server that receives requests intended for another server and that acts on behalf of the client (as the client proxy) to obtain the requested service. A proxy server is often used when the client and the server are incompatible for direct connection.

**PSTN** See *public switched telephone network*.

**public switched telephone network (PSTN)** The traditional analog network that routes voice calls from one location to another.

**query** A form of questioning in Structured Query Language (SQL) to retrieve information from a database.

**RDBMS** See *relational database management system*.

**Real Time Streaming Protocol (RTSP)** A client/server communication protocol that simplifies the distribution of multimedia contents on the Internet.

**Red Hat** A software company in the business of assembling open source components for the Linux operating system and related programs into a distribution package.

**relational database management system (RDBMS)** A type of DBMS in which the database is organized and accessed according to the relationships between data values.

**resilience** The ability of a structure to sustain the impact of a business interruption and recover and resume its operations to continue to provide minimum services.

**round robin** A simple mechanism used by Domain Name System (DNS) servers to share and distribute loads for network resources.

**RTSP** See *Real Time Streaming Protocol*.

**scalability** The ability to increase or decrease size or capability in cost-effective increments with minimal impact on the unit cost of business and the procurement of additional services.

**script** The set of instructions for the target voice response system to follow during a transaction.

**servlet** A server-side Java program that provides additional features to the server.

**software** The set or sets of programs that instruct the computer hardware to perform a task or series of tasks.

**Speech Synthesis Markup Language (SSML)** A markup language to control speech synthesis and text processing defined by the World Wide Web Consortium (W3C).

**SSML** See *Speech Synthesis Markup Language*.

**text to speech (TTS)** An optional feature that lets an application play language speech directly from ASCII text by converting that text to synthesized speech. The text can be used for prompts or for text retrieved from a database or host. And it can be spoken in an application with prerecorded speech.

**thin client** A low-cost computing device that accesses applications, data, or both from a central server over a network. Categories of thin clients include Windows-Based Terminals (WBT, which comprise the largest segment), X-Terminals, and Network Computers (NC).

**topology** The physical layout of a network.

**troubleshooting** The process of locating and correcting errors in computer programs.

**TTS** See *text to speech*.

**turnkey** A product or system that can be plugged in, turned on, and operated with little or no additional configuring.

**Uniform Resource Identifier (URI)** A formatted string used to locate via name or location a resource document such as a VoiceXML document.

**URI** See *Uniform Resource Identifier*.

**VoiceXtensible Markup Language (VoiceXML)** The W3C standard XML format for specifying interactive voice dialogues between a human and a computer. It is fully analogous to HTML. It brings the same advantages of Web application development and deployment to voice applications that HTML brings to visual applications.

**VoiceXML** See *Voice Extensible Markup Language*.

**WAR** See *Web archive*.

**Web archive (WAR)** The J2EE standard for packaging a Java servlet-based application, JavaServer™ Pages™, and associated files and deploying them.



# Abbreviations and acronyms

<b>ANI</b>	automatic number identification
<b>ASR</b>	automatic speech recognition
<b>CCXML</b>	Call Control Extensible Markup Language
<b>CTI</b>	Computer Telephony Integration
<b>DNIS</b>	dialed number identification service
<b>DTMF</b>	dual-tone multifrequency
<b>GUI</b>	graphical user interface
<b>HACMP</b>	High Availability Cluster Multi-Processing
<b>IBM</b>	International Business Machines Corporation
<b>ISDN</b>	Integrated Services Digital Network
<b>ITSO</b>	International Technical Support Organization
<b>J2EE</b>	Java 2 Platform, Enterprise Edition
<b>MRCP</b>	Media Resource Control Protocol
<b>NLSR</b>	Natural Language Speech Recognition
<b>PBX</b>	private branch exchange
<b>PSTN</b>	public switched telephone network
<b>RDBMS</b>	relational database management system
<b>RTSP</b>	Real Time Streaming Protocol
<b>SSI</b>	single system image
<b>SSML</b>	Speech Synthesis Markup Language
<b>TTS</b>	text to speech
<b>URI</b>	Uniform Resource Identifier
<b>VoiceXML</b>	Voice Extensible Markup Language



# Related publications

The publications listed in this section are considered particularly suitable for a more detailed discussion of the topics covered in this Redpaper.

## IBM Redbooks

For information about ordering these publications, see “How to get IBM Redbooks” on page 79. Note that some of the documents referenced here may be available in softcopy only.

- ▶ *IBM WebSphere Voice Server for Multiplatforms V5.1.1/V5.1.2 Handbook*, SG24-6447

## WebSphere Voice Response Documentation

- ▶ *WebSphere Edge Server for Multiplatforms Network Dispatcher (Load Balancer) Administration Guide*, GC31-8496-06
- ▶ *WebSphere Voice Response for AIX: Using the CCXML Browser*, GC34-6368
- ▶ *WebSphere Voice Response for AIX: Developing Java applications*, GC34-6377
- ▶ *WebSphere Voice Response for AIX: Deploying and Managing VoiceXML and Java Applications*, GC34-6378
- ▶ *WebSphere Voice Response for AIX: General Information and Planning*, GC34-6379
- ▶ *WebSphere Voice Response for AIX: Installation*, GC34-6380
- ▶ *WebSphere Voice Response for AIX: Problem Determination*, GC34-6382
- ▶ *WebSphere Voice Response for AIX: Voice over IP using Session Initiation Protocol*, GC34-6383
- ▶ *WebSphere Voice Response for AIX: Fax using Brooktrout*, GC34-6385
- ▶ *WebSphere Voice Response for AIX: Configuring the System*, SC34-6381
- ▶ *WebSphere Voice Response for AIX: Managing and Monitoring the System*, SC34-6384
- ▶ *WebSphere Voice Response for AIX: User Interface Guide*, SC34-6386
- ▶ *WebSphere Voice Response for AIX: Application Development using State Tables*, SC34-6387
- ▶ *WebSphere Voice Response for AIX: Designing and Managing State Table Applications*, SC34-6388
- ▶ *WebSphere Voice Response for AIX: Custom Servers*, SC34-6389
- ▶ *WebSphere Voice Response for AIX: 3270 Servers*, SC34-6390
- ▶ *WebSphere Voice Response for AIX: Cisco ICM Interface User's Guide*, SC34-6391
- ▶ *WebSphere Voice Response for AIX: Programming for the ADSI Feature*, SC34-6393
- ▶ *WebSphere Voice Response for AIX: Programming for the Signaling Interface*, SC34-6392

## Online resources

These Web sites and URLs are also relevant as further information sources:

- ▶ Deploying and Managing VoiceXML and Java Applications  
<http://publibfp.boulder.ibm.com/epubs/pdf/c3463781.pdf>
- ▶ Microsoft Window Server 2003 TechCenter  
<http://www.microsoft.com/technet/prodtechnol/windowsserver2003/default.msp>
- ▶ Red Hat Enterprise Linux server  
<http://www.redhat.com/software/rhel>
- ▶ Softel Home Page  
<http://www.softel.com>
- ▶ SUSE Linux Enterprise Server  
<http://www.novell.com/products/linuxenterpriseserver>
- ▶ World Wide Web Consortium (W3C) home page  
<http://www.w3c.org>
- ▶ W3C recommendation of Speech Recognition Grammar Specification (SRGS) V1.0  
<http://www.w3.org/TR/2004/REC-speech-grammar-20040316>
- ▶ W3C recommendation of Speech Synthesis Markup Language (SSML) V1.0  
<http://www.w3.org/TR/2004/REC-speech-synthesis-20040907>
- ▶ W3C working draft of Semantic Interpretation for Speech Recognition (SISR)  
<http://www.w3.org/TR/2003/WD-semantic-interpretation-20030401>
- ▶ W3C recommendation for VoiceXtensible Markup Language (VoiceXML) V2.0  
<http://www.w3.org/TR/voicexml20>
- ▶ WebSphere Application Server Zone  
<http://www.ibm.com/developerworks/websphere/zones/was>
- ▶ WebSphere Application Server V5.1.x Information Center  
[http://publib.boulder.ibm.com/infocenter/ws51help/index.jsp?topic=/com.ibm.websphere.base.doc/info/welcome\\_base.html](http://publib.boulder.ibm.com/infocenter/ws51help/index.jsp?topic=/com.ibm.websphere.base.doc/info/welcome_base.html)
- ▶ WebSphere Application Server Network Deployment V5.1.x Information Center  
[http://publib.boulder.ibm.com/infocenter/ws51help/index.jsp?topic=/com.ibm.websphere.nd.doc/info/welcome\\_nd.html](http://publib.boulder.ibm.com/infocenter/ws51help/index.jsp?topic=/com.ibm.websphere.nd.doc/info/welcome_nd.html)
- ▶ WebSphere Application Server Base and Network Deployment editions recommended updates  
<http://www.ibm.com/support/docview.wss?rs=180&uid=swg27004980>
- ▶ WebSphere Voice family product documentation library  
<http://www.ibm.com/developerworks/websphere/zones/voice/proddoc.html#wvs>
- ▶ WebSphere Voice Response for AIX family product documentation library  
[http://www-128.ibm.com/developerworks/websphere/zones/voice/vresponse\\_aix.html#1](http://www-128.ibm.com/developerworks/websphere/zones/voice/vresponse_aix.html#1)
- ▶ WebSphere Voice Response for AIX features and benefits  
[http://www.ibm.com/software/pervasive/voice\\_response\\_aix/about](http://www.ibm.com/software/pervasive/voice_response_aix/about)
- ▶ WebSphere Voice Response for AIX support page  
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- ▶ WebSphere Voice Response for AIX system requirements  
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- ▶ WebSphere Voice Server Education and class information  
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- ▶ WebSphere Voice Server for Multiplatforms V5.1.x Information Center  
<http://publib.boulder.ibm.com/infocenter/pvcvoice/51x/index.jsp>
- ▶ WebSphere Voice Server support page  
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# IBM WebSphere Voice Server for Multiplatforms V5.1.1/V5.1.2 and WebSphere Voice Response V4.2

## An Interoperability Guide



**Use WebSphere Voice Server Connector for interoperability**

**Apply best practices to VoiceXML development**

**Test and debug VoiceXML applications**

IBM WebSphere Voice Server V5.1.1 and V5.1.2 products leverage the power of IBM WebSphere Application Server, the IBM flagship product and leader in the industry. WebSphere Voice Server V5.1 is the first release of a novel (patent pending) architecture for speech servers, based on WebSphere Application Server. Add the power of the Java 2 Enterprise Edition platform, including the runtime, services, and container management, and the result is a powerful speech server that integrates into existing IT infrastructures.

WebSphere Voice Response for AIX V4.2, with the WebSphere Voice Server Connector software, delivers next generation voice processing that meets advanced call-routing and voice self-service needs for a contact center enterprise. It uses standard IBM Quad Digital Trunk Telephony Adapter (DTTA) or Digital Trunk Extended Adapter (DTEA) hardware for telephony support and a VoiceXML V2.1 compliant browser for application programming interfaces.

This IBM Redpaper is a guide for integrating IBM WebSphere Voice Server V5.1.1 and V5.1.2 and WebSphere Voice Response for AIX V4.2. It explains how to tailor and configure WebSphere Voice Server and WebSphere Voice Response in simple and complex topologies. It shows how to implement and deploy a simple voice-enabled application and presents best practices as applied to the Voice User Interface design. In addition, this paper explains how to develop, test, and deploy a simple voice-enabled application using VoiceXML. It demonstrates how to use automatic speech recognition and text-to-speech voice technologies.

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