Configuring Audio File Properties for Tcl IVR and VoiceXML Applications

This chapter explains how to configure audio file properties for Tcl IVR and VoiceXML applications.

For more information about this and related Cisco IOS voice features, see the following:

- "Overview of Cisco IOS Tcl IVR and VoiceXML Applications" on page 1
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm.

Note For releases prior to Cisco IOS Release 12.3(14)T, see the previous version of the *Cisco IOS Tcl IVR and VoiceXML Application Guide* at: http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/tcl_leg/index.htm

Feature History for Tcl IVR and VoiceXML Applications

This chapter includes information about configuring audio for different Tcl IVR and VoiceXML application features. For a feature history of all Tcl IVR and VoiceXML features, see "Cisco IOS Tcl IVR and VoiceXML Feature List" on page 2.

Contents

When developing and configuring a voice application, use this chapter and refer to the *Cisco VoiceXML Programmer's Guide* or the *Tcl IVR API Version 2.0 Programmer's Guide*.

Prerequisites for Audio Files

- You must configure basic VoiceXML application functionality as described in "How to Configure Basic Functionality for a Tcl IVR or VoiceXML Application" on page 24.
- The Cisco gateway must have connectivity to the media server where the audio files used by the Tcl or VoiceXML application are stored.
- In Cisco IOS Release 12.2(11)T, to play or record audio files using an RTSP server on a VoIP call leg, use one of the following configuration options on the originating gateway to ensure that packets are not dropped:

Note This prerequisite is not required for Cisco IOS Release 12(2)11T1 and later.

  - Two different physical Ethernet interfaces, one for the RTSP server and one for the VoIP interface to the terminating gateway.
  - If the RTSP server and the terminating gateway share the same physical Ethernet interface, you must configure the **no ip redirects** command for Ethernet interface 0/0 on the originating gateway.

Restrictions for Audio Files

The following sections list restrictions for audio recording and playback features:

Recording and Playback Restrictions

- Playback of audio recordings directly from an ESMTP server is not supported.
- For ESMTP recording, only a single mailbox address is supported in the mailto URL.
- Final silence detection, which lets the gateway terminate a recording after a defined length of silence, is not supported for RTSP recording. The final silence detection feature is disabled by default; it must be enabled by using the final silence property in the VoiceXML document.
- The **vad** command must be configured in the VoIP dial peer when final silence detection is needed to terminate a voice recording. When using speech recognition, however, the **no vad** command is required, so final silence is not detected for recording on IP call legs.
- RTSP multicast sessions are not supported by the Cisco IOS RTSP client.
- Full editing features (for example, seek, pause, rewind, append) are not supported.
- Maximum duration of a recording stored in local memory on the Cisco gateway is limited to the amount of available free memory. Memory limits can also be configured on the gateway by using the **ivr record memory session** and **ivr record memory system** commands.
- For RAM recordings submitted using HTTP POST:
  - Only the chunked transfer method is supported.
  - Recordings are sent using enctype of "multipart/form-data."

  - The “audio/basic” enctype, which was supported in Cisco IOS Release 12.2(2)XB, is not supported in later releases of Cisco IOS software.
- For audio files streamed to an HTTP server:
  - Only the chunked transfer method is supported.
  - Audio files are sent using enctype of “multipart/form-data.”
- Volume control is supported for audio files played from memory or chunked transfer mode.
- Rate control is supported only for audio files played from memory or chunked transfer mode using the G.711 codec.
- The following restrictions apply to Cisco IOS Release 12.2(2)XB:
  - Audio recordings are stored in .au file format; .wav format is not supported.
  - Supported codecs for recording are G.711 u-law and G.723.1.
  - Supported codecs for audio playback are G.711 u-law, G.723.1, G.726 (Cisco AS5300), and G.729.
  - RTSP recording is not supported.
  - Rate control, volume control, and prompt timing information are not supported.
  - RAM recordings can be submitted to an external HTTP server using the POST method with an enctype of “audio/basic.”

Note A VoiceXML document written to support the “audio/basic” MIME type for <submit> in Cisco IOS Release 12.2(2)XB is not supported by later releases of Cisco IOS software. Later releases of Cisco IOS software support only the “multipart/form-data” type for <submit>.

Codec Restrictions

- Audio prompts played to an incoming VoIP call leg must use the same codec as the codec used for the VoIP call setup. If the codec of the audio file is different from the codec negotiated for the call, the audio prompt playout fails and an error is generated.
- Codec used for playing audio prompts must be used for the duration of the call, even after prompt playout is completed.
- For recording or playback over an IP call leg, the codec negotiated between the originating and terminating ends must match the codec specified in the VoiceXML document. If the codec of the audio file is different than the codec negotiated for the call, the recording or playback fails and an error is generated.
- The **voice-class codec** command is not supported in a dial peer that is configured with a VoiceXML application. Using the **voice-class codec** command results in a codec mismatch error when attempting a VoiceXML recording or prompt playout. You must use the **codec** command instead.
- GSM-EFR and G.728 codecs are not supported on the Cisco AS5350 or Cisco AS5400 for audio recording and playback.
- The GSMFR codec is not supported on the Cisco Integrated Services Routers (ISRs) for media recording.
- For a list of codecs that are supported for recording by platform and Cisco IOS release, see the “Codec Support for Audio Recording” section on page 5.

- Audio players might not recognize audio recordings made by the Cisco gateway using some codecs. The gateway can play back all audio files recorded by the gateway. For more information, see the "Audio File Formats Supported for Recording and Playback" section on page 5.

Information About Audio File Properties for Tcl and VoiceXML Applications

To configure audio files properties for Tcl and VoiceXML applications, you must understand the following concepts:

Audio File Playout Methods

Tcl and VoiceXML applications can be configured for incoming POTS or VoIP call legs to play announcements to the user and to request user input (digits). The gateway can also play back audio recordings made by VoiceXML applications. Audio files can be played toward both the PSTN side and the IP side of the call leg.

Tcl and VoiceXML applications can play out audio files by using the following playout methods from different locations:

- Memory
- HTTP, Flash, TFTP, or FTP Streamed
- RTSP Streamed
- TTS Streamed

Memory

The entire audio file is loaded into the gateway's memory and then played out to the appropriate call leg as needed. Memory-based prompts can be loaded from an HTTP server, or from Flash memory, a TFTP server, or an FTP server. Audio files can also be recorded into memory using VoiceXML recording capabilities, then played back from memory or submitted to an external HTTP server to become permanent audio files.

The amount of memory available to store audio prompts on the gateway can be configured by using the **ivr prompt memory** command.

Note Flash memory allows a limited number of entries, typically 32 on most platforms. For the specific Flash memory limits for your platform, refer to the platform-specific reference documentation listed in the "Additional References" section on page 5.

HTTP, Flash, TFTP, or FTP Streamed

The Cisco gateway can stream audio files from an external server to the appropriate call leg as needed. Loading an entire audio prompt into local memory before beginning playout can limit the length of audio prompts and impact memory resources. With streaming, pieces of a prompt are loaded into memory and then, if necessary, deleted after they are played to free up memory. The audio file is played out while it is being loaded into memory, with playback beginning as soon as a piece of the prompt is loaded.

Prompts can be streamed from an HTTP server or from Flash memory, a TFTP server, or an FTP server. With HTTP, each time a prompt is played, the HTTP caching system is checked, and the audio file is reloaded if necessary. The HTTP cached flag in the VoiceXML document specifies whether an audio file that is loaded into memory is safe to use again, and does not have to be deleted.

To enable the gateway to stream audio files during playout, use the **ivr prompt streamed** command. HTTP prompts are streamed by default, but HTTP streaming can be disabled by using the **no ivr prompt streamed http** command.

The amount of memory available to store audio prompts on the gateway can be configured by using the **ivr prompt memory** command. Performance is best when there is enough memory to store the entire audio file. If the **ivr prompt memory** command is set to a value smaller than the size of a streamed file, performance is not as good.

RTSP Streamed

An external Real Time Streaming Protocol (RTSP) server can stream audio to the appropriate call leg as needed. RTSP is an application-level protocol that controls the on-demand delivery of real-time data, such as the delivery of audio streams from an audio server. By implementing an RTSP client on the Cisco VoIP gateway, a voice application running on the gateway can connect calls with audio streams from an external RTSP server. Prompts from RTSP servers are always streamed during playback. RTSP saves memory on the gateway because it is packet-based. Unlike HTTP or TFTP streaming, for example, RTSP streaming does not read any part of the audio file into RAM.

Note When playing a series of short audio prompts, such as with dynamic prompts, nonstreaming might be more efficient; streaming playout can cause noticeable delays and impact voice quality.

TTS Streamed

An external speech synthesizer using MRCP can generate prompts. Requests to synthesize speech from text strings or audio segments are sent to the media server, which responds with a real-time audio stream.

Dynamic Prompts

Dynamic prompts are formed by the underlying system assembling small audio files and playing them out in sequence. This provides simple TTS operations, like playing numbers, dollar amounts, dates, and time. For example, dynamic prompts can inform the caller of how much time is left in their debit account, as in:

> “You have 15 minutes and 32 seconds of call time left in your account.”

This prompt is created using eight individual audio files. They are: youhave.au, 15.au, minutes.au, and.au, 30.au, 2.au, seconds.au, and leftinyouraccount.au. These audio files are assembled dynamically by the underlying system and played out as a single prompt.

The language and location of the audio files used for dynamic prompts can be specified in the Tcl script or VoiceXML document, or these parameters can be configured on the Cisco gateway by using the **service** and **package** commands in the application configuration.

Note When playing a series of short audio prompts, such as with dynamic prompts, non-streaming might be more efficient; streaming playout can cause noticeable delays and impact voice quality.

Tcl Language Modules for Dynamic Prompts

Each language uses a Tcl language module. The Tcl language module defines the list of TTS notations that the language supports. Cisco IOS software includes built-in language modules for Chinese, English, and Spanish. You can add support for new languages and new TTS notations by configuring a new Tcl language module on the gateway.

The Cisco IOS infrastructure interfaces with the Tcl language module to translate TTS notations supplied by the voice application into the specified language. Cisco IOS software translates TTS notations into the sequence of audio files according to the language structure. For example, English and French use different sequences for saying the date: the English language structure says the month first and then the day; the French language structure says the day first and then the month.

Note Language modules are not used by external TTS servers; they are used by Cisco IOS software to assemble a list of dynamic prompts.

New TTS notations for the Cisco IOS built-in languages, such as playing dates and times of day, can also be configured. For example, if you configure a new English Tcl language module, it overrides the built-in English Tcl language module during the translation. When completed, any voice application can use the new notations, and the Cisco IOS infrastructure recognizes and plays the audio accordingly.

Note Tcl language modules are not Tcl IVR scripts. They are pure Tcl scripts and any system on the Cisco gateway (Tcl IVR 1.0, 2.0, VoiceXML, MGCP) can use the configured language with little or no change to the Cisco IOS configuration.

For information on writing a new Tcl language module, refer to the *Cisco Pre-Paid Debitcard Multi-Language Programmer's Reference*.

For information on configuring a new language module on the gateway, see the "Specifying a New Language Module for Dynamic Prompts" section on page 7.

Volume and Rate Controls for Audio Prompts using VoiceXML

The volume of audio prompts can be adjusted during playback. Audio prompts that are played out from memory or through chunked transfer mode using the G.711 codec can also be sped up or slowed down. A VoiceXML variable contains the rate and duration of the last prompt that was played.

The rate and volume of prompts is controlled by using Cisco attributes in the VoiceXML document. For detailed information, refer to the *Cisco VoiceXML Programmer's Guide*.

How to Configure Audio File Properties for Applications

- Specifying a New Language Module for Dynamic Prompts, page 7
- Setting Language and Location of Audio Files for Dynamic Prompts, page 8
- Setting Memory Recording Limits, page 11 (optional)

- Verifying Prompt Playout, page 12 (optional)
- Configuring Audio Prompt Streaming, page 13 (optional)
- Modifying Codec Complexity on the Cisco 3600 Series, page 13 (optional)

Specifying a New Language Module for Dynamic Prompts

Cisco includes built-in language modules for Chinese, English, and Spanish. This section explains how to add support for new languages and new text-to-speech (TTS) notations by configuring a new Tcl language module on the gateway.

Note Before configuring a new language module on the gateway, you must first obtain or write a new Tcl language module and install it on a server that is accessible to the gateway. For information, see the *Cisco Pre-Paid Debitcard Multi-Language Programmer's Reference*.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **package** *package-name location*
5. **param** *parameter-name parameter-value*

DETAILED STEPS

Step 1 Enable privileged EXEC mode:

```
enable
Example: Router> enable
```

Enter your password if prompted.

Step 2 Enter global configuration mode:

```
configure terminal
Example: Router# configure terminal
```

Step 3 Enter application configuration mode to configure applications and services:

```
application

Example: Router(config)# application
```

Step 4 Enter service parameter configuration mode:

```
package package-name location

Example: Router(config-app)# package french
tftp://server-1/tftpboot/scripts/fr_translate.tcl
```

Step 5 Configure the parameter's value:

```
param parameter-name parameter-value

Example: Router(config-app-param)# param language fr
    param prefix fr
```

Verifying Configured Languages

To verify configured languages, perform the following steps.

SUMMARY STEPS

1. **show call application voice** *language*

DETAILED STEPS

Step 1 Use the **show call language voice** command to display information about a specific language.

This example shows parts of a Russian (*ru*) Tcl language module.

```
Router# show call language voice ru
Script Name : russian
       URL  : builtin:package_russian.C
       Type : Package
       State: Registered
       Life : Builtin
       Exec Instances: 10

     Parameters registered under russian namespace:
     name                 type  default value   description
     location             S     flash:          The URI of the audio files for this
language
     prefix               S     ru              The prefix of this language
     language             S     ru              The language code of this language
     index                S     -1              The index of this language

   Script Code Begin:
   --------------------------------

   Built in C Generic language translation package for dynamic prompt

   --------------------------------
```

Troubleshooting Tips

If language configuration is not successful, verify the following:

- The language package you are using is loaded.
- The parameters you are configuring are contained in the language package.

Setting Language and Location of Audio Files for Dynamic Prompts

The language and location of audio files that are used for dynamic prompts can be configured on the gateway or these parameters can be specified through properties in the VoiceXML document or Tcl script. For information on specifying the language and location of dynamic prompts by using

VoiceXML or Tcl properties, refer to the *Cisco VoiceXML Programmer's Guide* or *Tcl IVR API Version 2.0 Programmer's Guide*, respectively. This section explains how to configure the language and location of dynamic prompts through the gateway.

Note Specifying the language and location of dynamic prompts in a VoiceXML document or Tcl script takes precedence over the Cisco gateway configuration. Any value that is configured on the gateway is ignored if the same attribute is specified using a VoiceXML or Tcl property.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. service *service-name*
5. **paramspace** *language* **language** *prefix*
6. **paramspace** *language* **index** *number*
7. **paramspace** *language* **location** *location*

DETAILED STEPS

Step 1 Enable privileged EXEC mode:

```
    enable
Example: Router> enable
```

Enter your password if prompted.

Step 2 Enter global configuration mode:

```
    configure terminal
Example: Router# configure terminal
```

Step 3 Enter application configuration mode:

```
    application

Example: Router(config)# application
```

Step 4 Enter the parameter configuration mode for this application:

```
    service service-name
```

Step 5 Specify the language prefix and index number of the audio files that an application uses for dynamic prompts:

```
application
 service service-name
  paramspace language language prefix
  paramspace language index number
```

- *service-name*—Name of the Tcl or VoiceXML application.
- *language*—Name of the language package being configured. There are three built-in language packages: Chinese, English, and Spanish. Other languages may be supported by use a of Tcl language script.
- *number*—Number that identifies the language of the audio files used for dynamic prompts. (This number has no significance in VoiceXML applications. Enter any number.)

- *prefix*—Two-character code for the language:
  - Chinese: ch
  - English: en (default)
  - Spanish: sp
  - all three: aa

```
Example:
Router(config)# application
Router(config-app)# service vapp1
Router(config-app-param)# paramspace english language en
Router(config-app-param)# paramspace english index 1
```

Note When configuring the language using the **index** command, keep in mind that the software is hardcoded with digit 1 to represent the primary language and digit 2 to represent the secondary language.

Step 6 Specify the location of the audio files that an application uses for dynamic prompts:

```
application
  service service-name
    paramspace language location location
```

- *service-name*—Name of the Tcl or VoiceXML application.
- *language*—Name of the language package.
- *location*—URL of the directory that contains the language audio files used by the application, without filenames. Flash memory (flash) or a directory on a server (TFTP, HTTP, or RTSP) are all valid.

```
Example:
Router(config)# application
Router(config-app)# service vapp1
Router(config-app-param)# paramspace english location flash
```

This command specifies the location of the language audio files that are used for the dynamic prompts specified in Step 1. Use this command multiple times for a single application to allow up to four subdirectories for audio files.

The following example shows the creation of two subdirectories:

```
Router(config)# application
Router(config-app)# service vapp1
Router(config-app-param)# paramspace english location http://aufiles/en_dir/
Router(config-app-param)# paramspace spanish location http://aufiles/sp_dir/
```

Verifying Language and Location of Audio Files for Dynamic Prompts

Use the **show running-config** command to see if the audio file language and location are properly configured with the **application** commands, for example:

```
Router# show running-config
!
```

```
application
 service vapp1 flash:demo0.vxml
  paramspace english language en
  paramspace english location flash
!
```

Setting Memory Recording Limits

The Cisco IOS VoiceXML feature supports the VoiceXML 1.0 <record> element for recording speech to a choice of four destinations, including local memory on the Cisco gateway. You can set limits on the amount of memory allocated for audio files recorded to local memory. This section explains how to modify the maximum memory limit for a single recording session (call) or for all recording sessions combined.

Note This procedure only configures memory limits for recordings to local memory on the Cisco gateway. Recording limits are not configurable on the gateway for HTTP, RTSP, or SMTP recordings.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ivr record memory session** *kilobytes*
4. **ivr record memory system** *kilobytes*

DETAILED STEPS

Step 1 Enable privileged EXEC mode:

enable

```
Example: Router> enable
```

Enter your password if prompted.

Step 2 Enter global configuration mode:

configure terminal

```
Example: Router# configure terminal
```

Step 3 Set the maximum amount of memory that can be used for recording in a single VoiceXML interpreter session:

ivr record memory session *kilobytes*

- *kilobytes*—Memory size in kilobytes. Range is from 0 to 256,000. The default is 256.

```
Example: Router(config)# ivr record memory session 1000
```

This example sets the maximum recording memory used by a single session to 1 MB of memory. For G.711 recordings, which use 8 KB/sec, this allows 125 sec of recording to be stored for each session.

Step 4 Set the maximum amount of memory that can be used to store all audio files:

ivr record memory system *kilobytes*

- *kilobytes*—Memory size in kilobytes. Range is from 0 to 256,000. If 0 is configured, the recording function is disabled on the gateway. The default memory size is platform-specific:
  - Cisco 3640 and Cisco AS5300: 10,000 KB
  - Cisco 3660, Cisco AS5350, and Cisco AS5400: 20,000 KB

```
Example: Router(config)# ivr record memory system 15000
```

Troubleshooting Tips

Table 4-1 lists possible causes for audio recording and playout faiing and suggested actions.

Table 4-1 Audio Recording or Playout Fails

Possible Causes	Suggested Actions
VCWare version is not supported (Cisco AS5300 only).	Use the **show vfc version** command to verify that the router is using VCWare version 10.25 or later.
Codec of the audio file on an incoming IP call leg is different than the codec negotiated for the call.	Compare the codec specified in the VoiceXML document with the codec configured in the dial peer on the gateway. If there is a codec mismatch, the audio playout fails and an error is generated.
Audio files recorded with different codecs are included in the same <prompt> elements in the VoiceXML document.	Verify that audio files recorded with different codecs use different <prompt> elements in the VoiceXML document.
RTSP server and VoIP interface share the same Ethernet interface.	See the "Prerequisites for Audio Files" section on page 2.
If using a third-party media tool, the media tool does not support the format of recordings made by the Cisco gateway.	Playback the recording from the gateway or see the "Correction Utility for Audio File Headers" section on page 7.

Verifying Prompt Playout

SUMMARY STEPS

1. **show vfc** *slot* **version vcware**
2. Verify that audio prompts played to an incoming IP call leg use the same codec as the codec used for the VoIP call setup.
3. Verify that audio files recorded with different codecs use different <prompt> elements in the VoiceXML document.

DETAILED STEPS

Step 1 If playing prompts from the Cisco AS5300, verify that the router is using VCWare version 10.25 or later, by using the **show vfc version** command, for example:

```
Router# show vfc 2 version vcware

Voice Feature Card in Slot 2:
     VCware Version     :  10.25
     ROM Monitor Version:  1.2
     DSPware Version    :  1.0
     Technology         :  C542
```

Step 2 Verify that audio prompts played to an incoming IP call leg use the same codec as the codec used for the VoIP call setup. If the codec of the audio file is different than the codec negotiated for the call, the audio prompt playout fails and an error is generated.

Step 3 Verify that audio files recorded with different codecs use different <prompt> elements in the VoiceXML document.

Configuring Audio Prompt Streaming

Audio prompts can be streamed during playback to help free up memory resources. With streaming, smaller pieces of a prompt are loaded into memory and then deleted as they are used, instead of loading the entire prompt into memory before beginning playout. For more information, see the "Audio File Playout Methods" section on page 4.

The following command specifies whether audio prompts from selected media types are streamed during playback:

ivr prompt streamed {**all** | **flash** | **http** | **none** | **tftp**}

- **all**—Audio prompts from all URL types (HTTP, TFTP, Flash memory) are streamed during playback.
- **flash**—Audio prompts from flash memory are streamed during playback.
- **http**—Audio prompts from an HTTP URL are streamed during playback. This is the default value.
- **none**—No audio prompts (HTTP, TFTP, Flash memory) from any media type are streamed during playback.
- **tftp**—Audio prompts from an TFTP URL are streamed during playback.

```
Example: Router(config)# ivr prompt streamed flash
```

Configuring RTSP Live Streaming

RTSP live streaming is supported in Cisco IOS Release 15.0(1)M and later releases. To stream prompts from an RTSP server, use the **rtsp client timeout connect** command to set the number of seconds allowed for the router to establish a TCP connection to an RTSP server. With Cisco IOS Release 15.0(1)M and later releases, use the *cisco-maxtime* attribute of the <prompt> element to control the playout time for RTSP live streaming. If *cisco-maxtime* is zero or has no value set, the RTSP stream is played indefinitely.

Modifying Codec Complexity on the Cisco 3600 Series

Codec complexity affects the number of calls that can take place on the digital signal processor (DSP) interfaces. The greater the codec complexity, the fewer the calls that can be handled. Codec complexity is either medium or high. The default is medium. All medium-complexity codecs can also run in high-complexity mode, but fewer (usually half as many) channels are available per DSP. The value configured for codec complexity determines the choice of codecs that are available in the dial peers.

Codec complexity also determines whether the Cisco 3600 series supports a separate media stream for speech recognition, enabling the gateway to simultaneously perform speech synthesis or play audio files using a different codec. A separate RTP stream for speech recognition is supported on the Cisco 3660 only when the codec complexity is set to high. It is not supported for medium complexity codecs.

This section explains how to modify the codec complexity on the Cisco 3600 series.

For details on the number of calls that can be handled simultaneously using each of the codec standards, refer to the following resources:

- *Cisco IOS Release 12.2 Cross-Platform Release Notes*
- *Digital T1/E1 Packet Trunk Network Module* data sheet
- *Cisco IOS Voice Command Reference, Release 12.3 T*, entries for the **codec** and **codec complexity** commands.

Note On Cisco 3600 series routers with digital T1/E1 packet voice trunk network modules (NM-HDV), codec complexity cannot be configured if DS0 groups are configured. If no DS0 groups are configured, you can skip Steps 6 through 8 in the following procedure.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port:ds0-group-no*
4. **shutdown**
5. **exit**
6. **controller** {**t1** | **e1**} *slot/port*
7. **no ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type** *type*
8. **exit**
9. **voice-card** *slot*
10. **codec complexity** {**high** | **medium**}
11. **exit**
12. Repeat Step 6, then continue with step 13.
13. **ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type** *type*
14. **exit**
15. Repeat Step 3, then continue with step 16.
16. **no shutdown**

DETAILED STEPS

Step 1 Enable privileged EXEC mode:

```
enable
Example: Router> enable
```

Enter your password if prompted.

Step 2 Enter global configuration mode:

```
configure terminal
Example: Router# configure terminal
```

Step 3 Enter voice-port configuration mode:

```
voice-port slot/port:ds0-group-no
```

- *slot*—Backplane slot number where the voice module is installed. Valid entries are platform dependant, in the range of 0 to 6. Refer to the *Software Configuration Guide for Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers* for more information.
- *port*—Voice interface card location. Valid entries are 0 or 1.
- *ds0-group-no*— Identifies the DS0 group. Range is from 0 to 23 for T1, 0 to 30 for E1.

`Example: Router(config)# voice-port 0/1:23`

Step 4 Deactivate the voice port:

shutdown

`Example: Router(config-voiceport)# shutdown`

Step 5 Exit voice port configuration mode and return to global configuration mode:

exit

`Example: Router(config-voiceport)# exit`

Step 6 Enter controller configuration mode for the selected T1 or E1 controller:

controller {**t1** | **e1**} *slot/port*

- *slot*—Backplane slot number of the interface. Valid entries are platform-dependant, in the range of 0 to 6. Refer to the *Software Configuration Guide for Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers* for more information.
- *port*—Backplane port number of the interface. Valid entries are 0 or 1.

`Example: Router(config)# controller t1 0/1`

Step 7 Remove the related DS0 groups:

no ds0-group *ds0-group-no*

- *ds0-group-no*— Identifies the DS0 group. Range is from 0 to 23 for T1, 0 to 30 for E1.

`Example: Router(config-controller)# no ds0-group 0 timeslots 1-24 type e&m-wink-start`

Step 8 Exit controller configuration mode and return to global configuration mode:

exit

`Example: Router(config-controller)# exit`

Step 9 Enter voice card configuration mode for the card or cards in the slot specified:

voice-card *slot*

- *slot*—Slot number of the voice card. Valid entries are platform dependant, in the range of 0 to 6. Refer to the *Software Configuration Guide for Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers* for more information.

`Example: Router(config)# voice-card slot`

Note If any DSP voice channels are in the busy state, codec complexity cannot be changed. You can use the **show voice dsp** command to check the DSP voice channel activity.

Step 10 Specify codec complexity based on the codec standard:

codec complexity {**high** | **medium**}

Note If two WAN interface cards are installed, this command configures both cards at once.

- **high**—(Optional) Supports up to six voice or fax calls per DSP module (PVDM-12), using the codecs: G.723, G.728, G.729, G.729 Annex B, fax relay, or any of the medium complexity codecs.
- **medium**—Supports up to 12 voice or fax calls per DSP module (PVDM-12), using the codecs: G.711, G.726, G.729 Annex A, G.729 Annex A with Annex B, and fax relay. The default is **medium**.

This setting restricts the codecs available in dial peer configuration. All voice cards in a gateway must use the same codec complexity setting.

```
Example: Router(config-voice-card)# codec complexity high
```

Step 11 Exit voice card configuration mode and return to global configuration mode:

`exit`

```
Example: Router(config-voice-card)# exit
```

Step 12 Repeat Step 6, then continue with Step 13.

Step 13 Add the related DS0 groups:

`ds0-group` *`ds0-group-no`* **`timeslots`** *`timeslot-list`* **`type`** *`type`*

- *ds0-group-no*— Identifies the DS0 group. Range is from 0 to 23 for T1, 0 to 30 for E1.
- *timeslot-list*—Single time slot number, single range of timeslot numbers, or multiple ranges of timeslot numbers separated by commas. Range is from 1 to 24 for T1, 1 to 31 for E1 (time slot 16 is reserved for signaling).
- *type*—Signaling type of the telephony connection.

```
Example: Router(config-controller)# ds0-group 0 timeslots 1-24 type e&m-wink-start
```

Step 14 Exit controller configuration mode and return to global configuration mode:

`exit`

```
Example: Router(config-controller)# exit
```

Step 15 Repeat Step 3, then continue with Step 16.

Step 16 Activate the voice port:

`no shutdown`

```
Example: Router(config-voiceport)# no shutdown
```

Configuration Examples for Audio Files

Configuration examples of Cisco Tcl and VoiceXML applications are in the "Configuration Examples for Tcl IVR and VoiceXML Applications" section on page 71.

Where to Go Next

- To configure voice recording using a VoiceXML application, see "Configuring VoiceXML Voice Store and Forward" on page 1.
- To configure properties for speech recognition or speech synthesis, see "Configuring ASR and TTS Properties" on page 1.
- To configure a VoiceXML fax detection application, see "Configuring Fax Detection for VoiceXML" on page 1.
- To configure telephony call-redirect features for voice applications, see "Configuring Telephony Call-Redirect Features" on page 1.
- To configure session interaction for a Tcl IVR 2.0 application, see "Configuring Tcl IVR 2.0 Session Interaction" on page 1.
- To configure support for SIP and TEL URLs, see "Configuring SIP and TEL URL Support" on page 245.
- To monitor and troubleshoot voice applications, see "Monitoring and Troubleshooting Voice Applications" on page 1.

Additional References

- “” on page 1—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, Tcl IVR and VoiceXML features for that release
- “Overview of Cisco IOS Tcl IVR and VoiceXML Applications” on page 1—Describes underlying Cisco IOS Tcl IVR and VoiceXML technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance