



Configuring Analog Voice Ports

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Analog voice port interfaces connect routers in packet-based networks to analog two-wire or four-wire analog circuits in telephony networks. Two-wire circuits connect to analog telephone or fax devices, and four-wire circuits connect to PBXs. Connections to the PSTN central office (CO) are typically made with digital interfaces.

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Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Prerequisites for Configuring Analog Voice Ports

- Obtain two- or four-wire line service from your service provider or from a PBX.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.
- Install at least one other network module or WAN interface card to provide the connection to the network LAN or WAN.
- Establish a working IP and Frame Relay or ATM network. For more information about configuring IP, refer to the *Cisco IOS IP Configuration Guide*, Release 12.4.
- Install appropriate voice processing and voice interface hardware on the router. See the [Information About Analog Voice Hardware, page 2](#).
- Gather the following information about the telephony connection of the voice port:



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- Telephony signaling interface: FXO, FXS, or E&M
- Locale code (usually the country) for call progress tones
- If FXO, type of dialing: DTMF (touch-tone) or pulse
- If FXO, type of start signal: loop-start or ground-start
- If E&M, type: I, II, III, or V
- If E&M, type of line: two-wire or four-wire
- If E&M, type of start signal: wink, immediate, delay-dial

If you are connecting a voice-port interface to a PBX, it is important to understand the PBX's wiring scheme and timing parameters. This information should be available from your PBX vendor or the reference manuals that accompany your PBX.

**Note**

The slot and port numbering of interface cards differs for each of the voice-enabled routers. For the specific slot and port designations for your hardware platform, refer to the [Cisco Interface Cards Hardware Installation Guides](#). More current information may be available in the release notes for the Cisco IOS software you are using.

Information About Analog Voice Hardware

**Note**

For current information about supported hardware, refer to the release notes for the platform and Cisco IOS release being used.

- [Cisco 880 Series Routers, page 2](#)
- [Cisco 1750 Modular Router, page 3](#)
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Cisco 880 Series Routers

Beginning with Cisco IOS Release 12.4(15)XZ, the Cisco 880 series fixed router platforms support the implementation of analog (FXS/DID/FXO) and digital (BRI S/T) voice ports. The IAD881B, IAD881F, IAD888B, and IAD888F models support voice interface FXS or BRI. The IAD881F and IAD888F models have four FXS ports and the IAD881B and IAD888B models support two ports for ISDN BRI digital voice interface.

In the IAD881B and IAD888B models, the voice BRI interface presents an ISDN S/T interface to connect either to an NT1 terminating an ISDN telephone network (TE-side) or to a TE user device such as an ISDN telephone or PBX (NT-side). In the IAD881B and IAD888B models, the BRI interface is available as the primary voice interface and is intended to be connected to a PBX (network side trunk). All the voice interfaces are onboard though they are recognized as a 4-port FXS VIC and a 2-port BRI VIC in order to leverage existing voice drivers.

**Note**

If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.

The C881 and C888 SRST models automatically detect a failure occurring in the network and initiate a process to auto-configure the router. This process provides call-processing backup redundancy for the IP and FXS phones and helps to ensure that telephony capabilities stay operational. All the IP or analog phones hanging off of a telecommuter site are controlled by the headquarters office call control (Cisco Unified CallManager or CallManager Express). In case of a WAN failure, the telecommuter router allows all phones to re-register to it in SRST mode and allow all inbound and outbound dialing to be routed off to the PSTN (using back up FXO or BRI port). Upon restoration of WAN connectivity, the system automatically shifts call processing back to the primary Cisco Unified Call Manager cluster.

Cisco 1750 Modular Router

The Cisco 1750 modular router provides VoIP functionality and can carry voice traffic (for example, telephone calls and faxes) over an IP network. To make a voice connection, the router must have a supported voice interface card (VIC) installed. The Cisco 1750 router supports two slots for either WAN interface cards (WICs) or VICs and supports one VIC-only slot. For analog connections, two-port VICs are available to support FXO, FXS, and E&M signaling. VICs provide direct connections to telephone equipment (analog phones, analog fax machines, key systems, or PBXs) or to a PSTN.

Cisco2600 Cisco 3600 and Cisco 3700 Series Routers

The Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers are modular, multifunction platforms that combine dial access, routing, LAN-to-LAN services, and multiservice integration of voice, video, and data in the same device.

Voice network modules installed in Cisco 2600 series, Cisco 3600 series, or Cisco 3700 series routers convert telephone voice signals into data packets that can be transmitted over an IP network. The voice network modules have no connectors; VICs installed in the network modules provide connections to the telephone equipment or network. VICs work with existing telephone and fax equipment and are compatible with H.323 standards for audio and video conferencing.

For analog telephone connections, low-density voice/fax network modules that contain either one or two VIC slots are installed in the network module slots. Each VIC is specific to a particular telephone signaling interface (FXS, FXO, or E&M); therefore, the VIC determines the type of signaling on that module.

For more information, refer to the following:

- [Cisco 2600 Series Routers Hardware Installation Guide](#)
- [Cisco 3600 Series Routers Hardware Installation Guide](#)
- [Cisco Network Modules Hardware Installation Guide](#)
- [Cisco Interface Cards Installation Guide](#)

Cisco MC3810

To support analog voice circuits, a Cisco MC3810 must be equipped with an AVM, which supports six analog voice ports. When you install specific signaling modules known as analog personality modules (APMs), the analog voice ports may be equipped for the following signaling types in various combinations: FXS, FXO, and E&M. For FXS, the analog voice ports use an RJ-11 connector interface to connect to analog telephones or fax machines (two-wire) or to a key system (four-wire). For FXO, the analog voice

ports use an RJ-11 physical interface to connect to a CO trunk. For E&M connections, the analog voice ports use an RJ-1CX physical interface to connect to an analog PBX (two-wire or four-wire).

Optional high-performance voice compression modules (HCMs) can replace standard voice compression modules (VCMs) to operate according to the voice compression coding algorithm (codec) specified when the Cisco MC3810 concentrator is configured. The HCM2 provides four voice channels at high codec complexity and eight channels at medium complexity. The HCM6 provides 12 voice channels at high complexity and 24 channels at medium complexity. One or two HCMs can be installed in a Cisco MC3810, but an HCM may not be combined with a VCM in one chassis.

For more information, refer to the [Cisco MC3810 Multiservice Concentrator Hardware Installation Guide](#).

**Note**

For current information about supported hardware, refer to the release notes for the platform and Cisco IOS release you are using.

Basic Parameters on Analog FXO FXS or E and M Voice Ports

This section describes commands for basic analog voice port configuration.

All the data recommended in the [Prerequisites for Configuring Analog Voice Ports](#), page 1 should be gathered before you start this procedure.

If you are configuring a Cisco MC3810 that has HCMs, you should also configure the codec complexity by performing the tasks in the [Configuring Codec Complexity on the Cisco MC3810](#), page 9.

**Note**

If you have a Cisco MC3810 or Cisco 3660 router, the compand-type a-law command must be configured on the analog ports only. The Cisco 2660, Cisco 3620, and Cisco 3640 routers do not require the configuration of the compand-type a-law command. However, if you request a list of commands, the compand-type a-law command will display.

In addition to the basic voice port parameters described in this section, there are commands that allow voice port configurations to be fine-tuned. In most cases, the default values for fine-tuning commands are sufficient for establishing FXO and FXS voice port configurations. E&M voice ports are more likely to require some configuration. If it is necessary to change some of the voice port values to improve voice quality or to match parameters on proprietary PBXs to which you are connecting, use the commands in this section and also in the "Fine-Tuning Analog and Digital Voice Ports" chapter.

After the voice port has been configured, make sure that the ports are operational by performing the tasks described in the following chapters:

- "Verifying Analog and Digital Voice-Port Configuration"
- "Troubleshooting Analog and Digital Voice Port Configurations"

For more information on these and other voice port commands, refer to the *Cisco IOS Voice Command Reference*.

**Note**

The commands, keywords, and arguments that you are able to use may differ slightly from those presented here, based on your platform, Cisco IOS release, and configuration. When in doubt, use Cisco IOS command help to determine the syntax choices that are available.

Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules

The term *codec* stands for *coder-decoder*. A codec is a particular method of transforming analog voice into a digital bit stream (and vice versa) and also refers to the type of compression used. Several different codecs have been developed to perform these functions, and each one is known by the number of the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) standard in which it is defined. For example, two common codecs are the G.711 and the G.729 codecs. The various codecs use different algorithms to encode analog voice into digital bit-streams and have different bit rates, frame sizes, and coding delays associated with them. The codecs also differ in the type of perceived voice quality they achieve. Specialized hardware and software in the digital signal processors (DSPs) perform codec transformation and compression functions, and different DSPs may offer different selections of codecs.

Select the same type of codec as the one that is used at the other end of the call. For instance, if a call was coded with a G.729 codec, it must be decoded with a G.729 codec. Codec choice is configured in dial peers. For more information, refer to the "Dial Peer Configuration on Voice Gateway Routers" document.

Codec complexity refers to the amount of processing power that a codec compression method requires. The greater the codec complexity, the fewer the calls that the DSP interfaces can handle. 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 codec complexity value determines the choice of codecs that are available in the dial peers when the **codec** command has been configured. For details on the number of calls that can be handled simultaneously using each of the codec standards, refer to the entries for the **codec** and **codec complexity** commands in the *Cisco IOS Voice Command Reference*.

How to Configure Analog Voice Ports

- [Configuring Basic Parameters on Analog FXO FXS or E and M Voice Ports, page 5](#)
- [Configuring Codec Complexity on the Cisco MC3810, page 9](#)

Configuring Basic Parameters on Analog FXO FXS or E and M Voice Ports

Perform this task to configure basic parameters:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Do one of the following:
 - **voice-port** *slot / port*
4. Do one of the following:
 - **signal** {**loop-start** | **ground-start**}
5. **cptone** *locale*
6. **dial-type** {**dtmf** | **pulse**}
7. **operation** {**2-wire** | **4-wire**}
8. **type** {**1** | **2** | **3** | **5**}
9. Do one of the following:
 - **ring frequency** {**25** | **50**}
 -
 -
 - **ring frequency** {**20** | **30**}
10. **ring number** *number*
11. **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] | [**define** *pulse interval*]}
12. **description** *string*
13. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

Command or Action	Purpose
<p>Step 3 Do one of the following:</p> <ul style="list-style-type: none"> • voice-port <i>slot / port</i> <p>Example:</p> <pre>Router(config)# voice-port 1/0</pre> <p>Example:</p> <pre>voice-port slot/subunit/port</pre> <p>Example:</p> <pre>Router(config)# voice-port 1/0/0</pre>	<p>Enters voice-port configuration mode.</p> <p>Note The slash must be entered between slot and port.</p> <ul style="list-style-type: none"> • Valid entries vary by router platform; enter the show voice port summary command for available values. <p>Note For the Cisco 880 series platforms, the command syntax does not include a slot number, only the port is identified. If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.</p>
<p>Step 4 Do one of the following:</p> <ul style="list-style-type: none"> • signal {loop-start ground-start} <p>Example:</p> <pre>Router(config-voiceport)# signal ground-start</pre> <p>Example:</p> <pre>signal {wink-start immediate-start delay-dial}</pre> <p>Example:</p> <pre>Router(config-voiceport)# signal wink-start</pre>	<p>Selects the access signaling type to match that of the telephony connection you are making.</p> <p>Note Configuring the signal keyword for one voice port on a Cisco 2600 or Cisco 3600 series router VIC changes the signal value for both ports on the VIC.</p>
<p>Step 5 cptone <i>locale</i></p> <p>Example:</p> <pre>Router(config-voiceport)# cptone us</pre>	<p>Selects the two-letter locale for the voice call progress tones and other locale-specific parameters to be used on this voice port.</p> <ul style="list-style-type: none"> • Cisco routers comply with the ISO 3166 locale name standards. To see valid choices, enter a question mark (?) following the cptone command. • The default is us.

Command or Action	Purpose
<p>Step 6 dial-type { dtmf pulse }</p> <p>Example:</p> <pre>Router(config-voiceport)# dial-type dtmf</pre>	<p>(FXO only) Specifies the dialing method for outgoing calls.</p>
<p>Step 7 operation { 2-wire 4-wire }</p> <p>Example:</p> <pre>Router(config-voiceport)# operation 4-wire</pre>	<p>(E&M only) Specifies the number of wires used for voice transmission at this interface (the audio path only, not the signaling path).</p> <ul style="list-style-type: none"> The default is 2-wire.
<p>Step 8 type { 1 2 3 5 }</p> <p>Example:</p> <pre>Router(config-voiceport)# type 2</pre>	<p>(E&M only) Specifies the type of E&M interface to which this voice port is connecting. See Table 2 in the "Voice Port Configuration Overview" chapter for an explanation of E&M types.</p> <ul style="list-style-type: none"> The default is 1.
<p>Step 9 Do one of the following:</p> <ul style="list-style-type: none"> ring frequency { 25 50 } ring frequency { 20 30 } <p>Example:</p> <pre>Router(config-voiceport)# ring frequency 50</pre> <p>Example:</p> <pre>Router(config-voiceport)# ring frequency 30</pre>	<p>(FXS only) Selects the ring frequency, in hertz, used on the FXS interface. This number must match the connected telephony equipment and may be country-dependent. If the ring frequency is not set properly, the attached telephony device may not ring or it may buzz.</p> <ul style="list-style-type: none"> The keyword default is 25 on the Cisco 1750 router, Cisco 2600 and Cisco 3600 series routers; and 20 on the Cisco MC3810.
<p>Step 10 ring number <i>number</i></p> <p>Example:</p> <pre>Router(config-voiceport)# ring number 1</pre>	<p>(FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router.</p> <ul style="list-style-type: none"> The default is 1.

Command or Action	Purpose
<p>Step 11 <code>ring cadence</code> {[<code>pattern01</code> <code>pattern02</code> <code>pattern03</code> <code>pattern04</code> <code>pattern05</code> <code>pattern06</code> <code>pattern07</code> <code>pattern08</code> <code>pattern09</code> <code>pattern10</code> <code>pattern11</code> <code>pattern12</code>] [<code>define pulse interval</code>]}</p> <p>Example:</p> <pre>Router(config-voiceport)# ring cadence pattern01</pre>	<p>(FXS only) Specifies an existing pattern for ring, or defines a new one. Each pattern specifies a ring-pulse time and a ring-interval time.</p> <ul style="list-style-type: none"> The default is the pattern specified by the cptone locale that has been configured.
<p>Step 12 <code>description</code> <i>string</i></p> <p>Example:</p> <pre>Router(config-voiceport)# description 255</pre>	<p>Attaches a text string to the configuration that describes the connection for this voice port. This description appears in various displays and is useful for tracking the purpose or use of the voice port. <i>The string</i> argument is a character string from 1 to 255 characters in length.</p> <ul style="list-style-type: none"> The default is that there is no text string (describing the voice port) attached to the configuration.
<p>Step 13 <code>no shutdown</code></p> <p>Example:</p> <pre>Router(config-voiceport)# no shutdown</pre>	<p>Activates the voice port. If a voice port is not being used, shut the voice port down with the shutdown command.</p>

Configuring Codec Complexity on the Cisco MC3810

To configure codec complexity for analog voice ports on the Cisco MC3810 using High-Performance Compression Modules (HCMs), use the following commands:

SUMMARY STEPS

1. `enable`
2. `show voice dsp`
3. `configure terminal`
4. `voice-card 0`
5. `codec complexity {high | medium}`

DETAILED STEPS

Command or Action	Purpose
Step 1 <code>enable</code> Example: <pre>Router> enable</pre>	Enables privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2 <code>show voice dsp</code> Example: <pre>Router# show voice dsp</pre>	Checks the DSP voice channel activity. If any DSP voice channels are in the busy state, the codec complexity cannot be changed. When all the DSP channels are in the idle state, continue to Step 3.
Step 3 <code>configure terminal</code> Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 4 <code>voice-card 0</code> Example: <pre>Router(config)# voice-card 0</pre>	Enters voice-card configuration mode and specifies voice card 0.
Step 5 <code>codec complexity {high medium}</code> Example: <pre>Router(config-voicecard)# codec complexity high</pre>	Specifies codec complexity based on the codec standard being used. This setting restricts the codecs available in dial peer configuration. All voice cards in a router must use the same codec complexity setting. Note If two HCMs are installed, this command configures both HCMs at once.

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