Configuring the Cisco Fourth-Generation Voice and Fax Network Interface Module

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The following Cisco Fourth-Generation Voice & Fax Network Interface Modules provide voice and Unified Communications services on the Cisco 4400/4300 Series Integrated Services Router:

- NIM-2FXS
- NIM-4FXS
- NIM-2FXO
- NIM-4FXO
- NIM-2FXS/4FXO
- NIM-4E/M
- NIM-2BRI
- NIM-4BRI

Supported Features

- Support for Foreign Exchange Office (FXO) and Foreign Exchange Station (FXS).
- Support for recEive and transMit or Ear and Mouth (E&M) and Basic Rate Interface (BRI) analog ports.
- Support for Cisco Unified Communications Manager Express (CME) and Media Gateway Control Protocol (MGCP).
- Support for STC application supplementary services.

Note For a list of features not supported on the Cisco Fourth-Generation Voice & Fax Network Interface Module in the Cisco IOS XE Release 3.14S, see the Unsupported Features, on page 2 section.

Unsupported Features

Following is a list of features not supported on the Cisco Fourth-Generation Voice & Fax Network Interface Module in Cisco IOS XE Release 3.14S:

- Codecs: iLBC, iSAC, G.723.1
- Connection trunk
- Hoot and holler, voice multicasting
- Music on hold (MoH) from a live feed
- Noise Reduction (NR)
- Secure Cisco Unified CME
- Secure Cisco Unified SRST
- Signal LMR (under E&M port)
- SIP supplementary call features with analog phones
- Trunk connection for tie lines, nailed up calls

Restrictions

- Surprise OIR is not supported.
- Managed OIR is not supported with active calls.

To determine whether there are any active calls before proceeding with managed OIR, use a command such as **show voice call summary**. Ensure that all ports are in an "ONHOOK" state. After module insertion, check if the voice ports are in a shutdown state and issue **no shutdown** commands to bring each port back online.

Supported Platforms

The Cisco Fourth-Generation Voice & Fax Network Interface Module is supported on the Cisco 4451-X Integrated Services Router and runs on Cisco IOS XE Release 3.13S and later.

Configuring the Network Interface Module

Prerequisites for Configuring the Cisco Fourth-Generation Voice and Fax Network Interface Module

- 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 connection to the network.
- Install appropriate voice interface hardware on the router
- Gather the following information about the telephony connection of the voice port:
  - Telephony signaling interface: FXO and FXS
  - Locale code (usually the country) for call progress tones
  - For FXO, type of dialing: DTMF (touch-tone) or pulse and type of signal: loop-start or ground-start
- Disconnect signaling by performing the following set of tasks:
  - supervisory disconnect signal
  - battery-reversal
  - no supervisory disconnect signal. See Understanding FXO Disconnect Problem for detailed configuration information.

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

Configuring an FXO Interface

To configure the Cisco Fourth-Generation Voice & Fax Network Interface Module as an FXO interface, perform the following task.

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot*/*subunit*/*subslot* **Example:** **Example:** `Router(config)# voice-port 0/2/0`	Enters voice-port configuration mode.
Step 4	**signal** {**groundStart** | **loopStart**} **Example:** `Router(config-voiceport)# signal groundStart`	Selects the access signaling type to match that of the telephony connection that you are making. The default setting for FXO and FXS voice ports in **loopStart**. • groundStart — Specifies the use of groundstart signaling used for FXO and FXS interfaces. Groundstart signaling allows both sides of a connection to place a call and to hang up. • loopStart — Specifies the use of loop start signaling used for FXO and FXS interfaces. With loopstart signaling, only one side of a connection can hang up. **Note** The CAMA version of the keywords **groundStart** and **loopStart** are **groundstart** and **loopStart** respectively.
Step 5	cptone locale **Example:** **Example:** `Router(config-voiceport)# cptone us`	Selects the two-letter locale for the voice call progress tones and other locale-specific parameters to be used on this voice port. The default is us.
Step 6	dial-type **{dtmf** | **mf** | **pulse}** **Example:** `Router(config-voiceport)# dial-type dtmf`	Specifies the dialing method for outgoing calls. The default dialing method is **dtmf** touch-tone dialing. • **dtmf** — Specifies the dual tone multifrequency (DTMF) touch-tone dialing.

	Command or Action	Purpose
		• **mf** — Specifies the multifrequency tone dialing. • **pulse** — Specifies the pulse (rotary) dialing.
Step 7	ring number number **Example:** **Example:** `Router(config-voiceport)# ring number 1`	Specifies the maximum number of rings to be detected before an incoming call is answered by the router. The default is 1.
Step 8	description string **Example:** `Router(config-voiceport)# description Voice Port One`	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. The string argument is a character string from 1 to 255 characters in length. By default, there is no text string (describing the voice port) attached to the configuration.
Step 9	no shutdown **Example:** `Router(config-voiceport)# no shutdown`	Activates the voice port. If a voice port is not being used, shut down the voice port by using shutdown command.

Examples

The following example shows two options for configuring an FXO interface.

```
1)
voice-port 0/1/0
    st4451(config-voiceport)#secondary ?
  dialtone  Secondary dialtone option for FXO port
st4451(config-voiceport)#secondary dialtone ?
  <cr>
2) voice-port 0/1/0
      st4451(config-voiceport)#connection ?
  plar      Private Line Auto Ringdown
     st4451(config-voiceport)#connection plar ?
  WORD  A string of digits including wild cards
  opx   Off-Premises eXtension PLAR
st4451(config-voiceport)#connection plar opx ?
  WORD       A string of digits including wild cards
st4451(config-voiceport)#connection plar opx 2345
```

Configuring an FXS Interface

To configure the Cisco Fourth-Generation Voice & Fax Network Interface Module as an FXS interface, perform the following task.

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	voice-card 0/2 **Example:** `Router(config)# local-bypass`	Configures local bypass on the voice-port. When a call is made across two different ports under the same FXS voice-cards, configuring the **local bypass** command allows the call to bypass the backplane and DSPs. However, when call is made across voice-ports belonging to two different voice-cards, the NIM card DSPs are invoked irrespective of the **local bypass** configuration.
Step 4	**voice-port** *slot/subunit/subslot* **Example:** `Router(config)# voice-port 0/2/0`	Enters voice-port configuration mode.
Step 5	**signal {did {delay-dial | immediate | loopStart} | groundStart | loopStart}** **Example:** `Router(config-voiceport)# signal groundStart`	Selects the access signaling type to match that of the telephony connection that you are making. **Note** Cisco IOS XE Release 3.13S supports only **groundStart** and **loopStart** signaling types.
Step 6	cptone locale **Example:** **Example:** `Router(config-voiceport)# cptone us`	Selects the two-letter locale for the voice call progress tones and other locale-specific parameters to be used on this voice port. The default is **us.**
Step 7	ring frequency {20 | 25 | 30 | 50} **Example:** `Router(config-voiceport)# ring frequency 50`	Selects the ring frequency, in hertz, used on the FXS interface. The frequency 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.
Step 8	Do one of the following: • ring cadence {*pattern-number* |[define pulse interval]} **Example:**	Specifies an existing ring pattern or defines a new one. The following ring cadence patterns have a predefined ring-pulse time and a ring-interval time.

	Command or Action	Purpose

	`Router(config-voiceport)# ring cadence pattern01`
	Example:
	
	Example:
	Example:
	
	`Router(config-voiceport)# ring cadence define 2 4 3 1`
		• pattern01—2 seconds on, 4 seconds off
		• pattern02—1 second on, 4 seconds off
		• pattern03—1.5 seconds on, 3.5 seconds off
		• pattern04 —1 second on, 2 seconds off
		• pattern05—1 second on, 5 seconds off
		• pattern06 —1 second on, 3 seconds off
		• pattern07— 0.8 second on, 3.2 seconds off
		• pattern08—1.5 seconds on, 3 seconds off
		• pattern09—1.2 seconds on, 3.7 seconds off
		• pattern10—1.2 seconds on, 4.7 seconds off
		• pattern11—0.4 second on, 0.2 second off, 0.4 second on, 2 seconds off
		• pattern12—0.4 second on, 0.2 second off, 0.4 second on, 2.6 seconds off
		
		The default is the pattern specified by the cptone locale that has been configured.
		
		• **define**—User-definable ring cadence pattern.Each number pair specifies one ring-pulse time and one ring-interval time. You must enter numbers in pairs, and you can enter from 1 to 6 pairs. The second number in the last pair that you enter specifies the interval between rings.

Step 9	description string **Example:** `Router(config-voiceport)# description Voice Port One`	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. The *string* argument is a character string from 1 to 255 characters in length. By default, there is no text string (describing the voice port) attached to the configuration.
Step 10	no shutdown **Example:** `Router(config-voiceport)# no shutdown`	Activates the voice port. If a voice port is not being used, shut down the voice port by using the shutdown command

Configuration Examples

The following example shows a partial running configuration of an FXS interface.

```
voice-card 0/2
# using default local-bypass
!
voice-port 0/2/0
 cptone CA
!
voice-port 0/2/1
 signal groundStart
```

```
!
voice-port 0/2/2
 signal did loop-start
 cptone CA
!
voice-port 0/2/3
connection plar 12345
dial-peer voice   20 pots
 destination pattern   33020
 port 0/2/0
dial-peer voice   21 pots
 destination pattern   33021
 port 0/2/1
dial-peer voice   22 pots
 destination pattern   33022
 port 0/2/2
dial-peer voice   23 pots
 destination pattern   33023
 port 0/2/3
dial-peer voice   12345  voip
 destination pattern   12345
 session target ipv4:1.5.25.100
```

The following example shows a partial running configuration of an FXO interface.

```
voice-card 0/3
 no  local-bypass
!
voice-port 0/3/0
 cptone CA
 connection plar opx 12345
!
voice-port 0/3/1
 signal groundStart
 connect plar  12345
!
voice-port 0/3/2
 secondary dialtone
 cptone CA
!
voice-port 0/2/3
 connect plar 12345
dial-peer voice   30 pots
 destination pattern   33030
 port 0/3/0
dial-peer voice   31 pots
 destination pattern   33031
 port 0/3/1
dial-peer voice   32 pots
 destination pattern   33032
 port 0/3/2
dial-peer voice   23 pots
 destination pattern   33033
 port 0/3/3
dial-peer voice   12345  voip
 destination pattern   12345
 session target ipv4:1.5.25.100
```

Configuring an E and M Interface

To configure the Cisco Fourth-Generation Voice & Fax Network Interface Module as an E&M interface, perform the following task.

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot*/*subunit*/*subslot* **Example:** `Router(config)# voice-port 0/2/0`	Enters voice-port configuration mode.
Step 4	**signal {wink-start | immediate-start | delay-dial}** **Example:** `Router(config-voiceport)# signal wink-start`	The keywords are as follows: • wink-start—(default) Indicates that the calling side seizes the line, then waits for a short off-hook wink from the called side before proceeding. • immediate-start—Indicates that the calling side seizes the line and immediately proceeds; used for E&M tie trunk interfaces. • delay-dial—Indicates that the calling side seizes the line and waits, then checks to determine whether the called side is on-hook before proceeding; if not, it waits until the called side is on-hook before sending digits. Used for E&M tie trunk interfaces.
Step 5	cptone locale **Example:** `Router(config-voiceport)# cptone us`	Selects the two-letter locale for the voice call progress tones and other locale-specific parameters to be used on this voice port. The default is **us.**
Step 6	**operation {2-wire | 4-wire}** **Example:** `Router(config-voiceport)# operation 4-wire`	Specifies the number of wires used for voice transmission at this interface (the audio path only, not the signaling path). The default is 2-wire.
Step 7	**type {1 | 2 | 3 | 5}** **Example:** `Router(config-voiceport)# type 2`	Specifies the type of E&M interface to which this voice port is connecting. See Table 5 for an explanation of E&M types. The default is 1.
Step 8	description string **Example:**	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

	Command or Action	Purpose
	`Router(config-voiceport)# description Voice Port One`	use of the voice port. The *string* argument is a character string from 1 to 255 characters in length. By default, there is no text string (describing the voice port) attached to the configuration.
Step 9	no shutdown **Example:** `Router(config-voiceport)# no shutdown`	Activates the voice port. If a voice port is not being used, shut down the voice port by using the shutdown command

Configuration Examples

The following example shows a partial running configuration of an E&M interface.

```
1) Select the signal  protocol
st4451(config-voiceport)#signal ?
  delay-dial  delay before dialing
  immediate   start immediately
  wink-start  start upon wink (default)
2) Specify the E&M interface type
st4451(config-voiceport)#type ?
  1  E&M type I (default)
  2  E&M type II
  3  E&M type III
  5  E&M type V
3) Specify the operation of the E&M signal
        st4451(config-voiceport)#operation ?
  2-wire  2-wire operation (default)
  4-wire  4-wire operation
voice-port 0/3/0
operation 4-wire
type 2
```

Configuring a BRI Interface

To configure the Cisco Fourth-Generation Voice & Fax Network Interface Module as a BRI interface, perform the following task:

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**isdn switch-type** *switch-type*	Configures the telephone company ISDN switch type.

	Command or Action	Purpose
	Example: `Router(config)# isdn switch-type basic-net3`	**Note** The BRI switch types that are supported are: net3 and qsig.
Step 4	**interface bri** *slot/subslot/port*: 0 **Example:** `Router(config)# interface bri 0/1/0:0`	Enter interface configuration mode to configure parameters for the specified interface. • *slot*—Slot location in which the BRI module resides (0 to 4). • *subslot*—Subslot location in which the BRI module resides (1 to 3). • *port*—Port number of the BRI module (0 to 3).
Step 5	**no ip address** **Example:** `Router(config-if)# no ip address`	Specifies that there is no IP address for this interface.
Step 6	**isdn overlap-receiving** **Example:** `Router(config-if)# isdn overlap-receiving`	(Optional) Activates overlap signaling to send to the destination PBX. In this mode, the interface waits for possible additional call-control information.
Step 7	**isdn** spid2 spid-number [ldn] **Example:** `Router(config-if)# isdn spid2 spid-number 415988488202`	(Optional; TE only) Specifies a SPID and optional local directory number for the B2 channel.
Step 8	shutdown **Example:** `Router(config-if)# shutdown`	Turns off the port (prior to setting the port emulation).
Step 9	isdn layer1-emulate {user | network} **Example:** `Router(config-if)# isdn layer1-emulate network`	Configures the Layer 1 port mode emulation and clock settings. • user—Configures the port as TE and sets it to function as a clock slave. This is the default. • network—Configures the port as NT and sets it to function as a clock master.
Step 10	no shutdown **Example:** `Router(config-if)# no shutdown`	Turns on the port.
Step 11	isdn protocol-emulate {user | network} **Example:**	Configures the Layer 2 and Layer 3 port protocol emulation. The keywords are as follows:

	Command or Action	Purpose
	`Router(config-if)# isdn protocol-emulate network`	• user—Configures the port as TE; the PBX is the master. This is the default. • network—Configures the port as NT; the PBX is the slave.
Step 12	isdn sending-complete **Example:** `Router(config-if)# isdn sending-complete`	(Optional) Configures the voice port to include the "Sending Complete" information element in the outgoing call setup message. This command is used in some geographic locations, such as Hong Kong and Taiwan, where the "Sending Complete" information element is required in the outgoing call setup message.
Step 13	isdn static-tei tei-number **Example:** `Router(config-if)# isdn static-tei 0`	(Optional) Configures a static ISDN Layer 2 terminal endpoint identifier (TEI). The value of tei-number can be from 0 to 64.
Step 14	**isdn point-to-point-setup** **Example:** `Router(config-if)# isdn point-to-point-setup`	(Optional) Configures the ISDN port to send SETUP messages on the static TEI. **Note** A static TEI must be configured in order for this command to be effective.
Step 15	**end** **Example:** `Router(config-if)# end`	Exits interface configuration mode.
Step 16	clear interface bri *slot/subslot/port:0* **Example:** `Router# clear interface bri 0/1/0:0`	(Optional) Resets the specified interface. The interface needs to be reset if the static TEI number has been configured in Step 16. • *slot*—Slot location in which the BRI module resides (0 to 4). • *subslot*—Subslot location in which the BRI module resides (1 to 3). • *port*—Port number of the BRI module (0 to 3).

Configuration Examples

The following example shows a partial running configuration of a BRI interface.

```
interface BRI0/1/0:0
 isdn switch-type basic-net3
 isdn protocol-emulate network
 isdn point-to-point-setup
 isdn layer1-emulate network
 isdn skipsend-idverify
!
interface BRI0/1/1:0
 isdn switch-type basic-net3
 isdn point-to-point-setup
```

```
 isdn skipsend-idverify
!
interface BRI0/1/2:0
 isdn switch-type basic-qsig
 isdn point-to-point-setup
 isdn skipsend-idverify
!
interface BRI0/1/3:0
 isdn switch-type basic-qsig
 isdn protocol-emulate network
 isdn point-to-point-setup
 isdn layer1-emulate network
 isdn skipsend-idverify
dial-peer voice 100 pots
 destination-pattern 100
 direct-inward-dial
 forward-digits all
 port 0/1/0
```

```
dial-peer voice 200 pots
 destination-pattern 200
 direct-inward-dial
 forward-digits all
 port 0/1/1
dial-peer voice 300 pots
 destination-pattern 300
 direct-inward-dial
 forward-digits all
 port 0/1/2
dial-peer voice 400 pots
 destination-pattern 400
 direct-inward-dial
 forward-digits all
 port 0/1/3
```

Media Gateway Control Protocol

Media Gateway Control Protocol (MGCP) defines a centralized architecture for creating multimedia applications, including Voice over IP (VoIP) See the Cisco IOS MGCP and Related Protocols Configuration Guide .

The Cisco ISRs are configured primarily as residential gateways (RGWs) under MGCP. For residential gateway configuration information, see the " Configuring an RGW " section of the "Basic MGCP Configuration" chapter of the Cisco IOS MGCP and Related Protocols Configuration Guide .

Configuring Cisco Unified CME

Cisco Unified Communications Manager Express is a feature-rich, entry-level IP telephony solution that is integrated directly into Cisco IOS software. Cisco Unified CME allows small business customers and autonomous small enterprise branch offices to deploy voice, data, and IP telephony on a single platform for small offices, thereby streamlining operations and lowering network costs.

Cisco Unified CME is ideal for customers who have data connectivity requirements and also need a telephony solution in the same office. Whether offered through a service provider's managed services or purchased directly by a corporation, Cisco Unified CME offers most of the core telephony features required in the small office and also many advanced features not available with traditional telephony solutions. The ability to deliver IP telephony and data routing using a single converged solution allows customers to optimize their operations and maintenance costs, resulting in a very cost-effective solution that meets office needs.

A Cisco Unified CME system is extremely flexible because it is modular. A Cisco Unified CME system consists of a router that serves as a gateway and one or more VLANs that connect IP phones and phone devices to the router.

For more information on Cisco Unified CME, see the Cisco Unified Communications Manager Express System Administrator Guide .

Supported Cisco Unified Communications Manager Release for FXS, FXO, and BRI NIMs

The following table shows the Cisco Unified Communications Manager releases that are required to support FXS, FXO and BRI NIMs on the ISR 4000 series.

Table 1: Supported Cisco Unified Communications Manager Release

Cisco ISR 4000 Series	Cisco Unified Communications Manager Release
ISR 44*xx*	10.5
ISR 43*xx*	10.5.2

STC Application Supplementary Services

The SCCP telephony control (STC) application on the Cisco 4400 Series ISR functions as a proxy to translate call-control messages between the Cisco call-control system and the voice gateway. The SCCP telephony control (STC) application on the Cisco voice gateway presents the locally attached analog telephones as individual endpoints to the call-control system, which allows the analog phones to be controlled in the same way as IP phones. With this capability, gateway-attached endpoints share the same telephony features that are available on IP phones directly connected to Cisco Unified CME and Ciso unified Communications Manager.

Calls through analog FXS ports are controlled by a Cisco call-control system, such as Cisco Unified Communications Manager or Cisco Unified CME. The SCCP telephony control (STC) application on the Cisco voice gateway functions as a proxy to translate call-control messages between the Cisco call-control system and the Cisco voice gateway. See the Overview of Supplementary Services Features for FXS Ports on Cisco Voice Gateways for more information.

Troubleshooting

Use the following commands to check the status and troubleshoot the modules.

- debug mgcp packets
- **debug vpm sig**
- **debug voip vtsp default**
- show ccm-manager
- **show controller**
- show call active voice
- show call history voice
- show dial-peer voice summary
- show dialplan number
- **show hw-module subslot**
- **show interface serial**
- **show interface**
- show mgcp
- show mgcp connection

- show mgcp statistics
- **show platform hardware subslot (4400)**
- show voice call summary
- show voice call status
- show voice dsp
- show voice dsp channel operational-status
- show voice port
- show voice port summary
- show voice port 0/3/0 (example port)

Related Documents

Related Topic	Document Title
Installation guide for the Cisco PVDM4	Installing the Cisco PVDM4
Installation guide for the Cisco Network Interface Module	Installing the Cisco Fourth-generation Voice and WAN Network Interface Module
Command reference information for interface and hardware components	Cisco IOS Interface and Hardware Component Command Reference
Configuration of the Cisco 4400/4300 Series Integrated Services Router	Cisco 4400 Series ISRs and Cisco 4300 Series ISRs Software Configuration Guide
Installation of the Cisco 4400/4300 Series Integrated Services Router	Hardware Installation Guide for the Cisco ISR 4400 and Cisco ISR 4300 Series Integrated Services Router
System administrator's guide for Cisco Unified SRST	Cisco Unified SCCP and SIP SRST System Administrator Guide (All Versions)
MGCP and Related Protocols Configuration Guide	Cisco IOS MGCP and Related Protocols Configuration Guide
MGCP Gateway Verification and Troubleshooting	Verify and Troubleshoot the Cisco IOS MGCP Gateway
Regulatory compliance and safety information	Cisco Network Modules and Interface Cards Regulatory Compliance and Safety Information

MIBs

MIB	MIBs Link
• CISCO ENTITY MIB • CISCO-ENTITY-ALARM-MIB • CISCO-ENTITY-SENSOR-MIB • CISCO-SIP-UA-MIB • CISCO-SYSLOG-MIB • CISCO-VOICE-ANALOG-IF-MIB • CISCO-VOICE-DIAL-CONTROL-MIB • CISCO-VOICE-IF-MIB • ENTITY-MIB • IF-MIB	To locate and download MIBs for selected platforms, Cisco software releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 1315	*Management Information Base for Frame Delay DTEs*
RFC 1406	Definitions of Managed Objects for the DS1 and E1 Interface Types

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