



# Voice Enhancement Features for Cisco IOS Release 12.4(24)T

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This chapter describes feature support that has been added in Cisco IOS Release 12.4(24)T on Cisco VGD 1T3, Cisco AS5350XM, and AS5400XM voice gateways. These features enhance performance of the voice gateways.

## **Finding Feature Information in This Module**

Your Cisco IOS software release may not support all of the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To reach links to specific feature documentation in this chapter and to see a list of the releases in which each feature is supported, use the “[Information About Voice Enhancement Features for Cisco IOS Release 12.4\(24\)T](#)” section on page 186.

## **Finding Support Information for Platforms and Cisco IOS and Catalyst OS Software Images**

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

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## Prerequisites for Voice Enhancement Features for Cisco IOS Release 12.4(24)T

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- To enable Secure Real-Time Transport Protocol (SRTP) on a Cisco VGD 1T3, Cisco AS5350XM, or Cisco AS5400XM configured as a Media Gateway Control Protocol (MGCP) gateway, first establish an IPsec connection between the Cisco Unified Communications Manager and the MGCP gateway before using the MGCP SRTP package. Otherwise, media keys are sent in clear text and your voice call is not secure.

## Restrictions for Voice Enhancement Features for Cisco IOS Release 12.4(24)T

- For the Private Line Automatic Ringdown (PLAR) Connection feature, only E&M immediate start signaling configured on the T1 within the CT3 line is supported. This feature does not support Foreign Exchanger Service (FXS) loop-start signaling.
- The Media Forking feature is not supported on the Cisco Unified Border Element, and SRTP is not supported on the forked stream.

## Information About Voice Enhancement Features for Cisco IOS Release 12.4(24)T

To configure these enhancement features, you should understand the following:

- [SIP Support for Media Forking, page 187](#)
- [Automatic Switch to G.Clear Codec when ISDN Bearer Capability IE Indicates 64K Unrestricted Bearer, page 187](#)
- [Private Line Automatic Ringdown for Trading Turrets Connection, page 187](#)
- [Connection Trunk and Transparent CCS for Trading Turrets, page 188](#)
- [Modem Pass-through Using SIP re-invite, page 188](#)
- [Voice Port Bit Conditioning, page 190](#)
- [Voice Trunk Conditioning and Trunk Conditioning OOS, page 190](#)
- [Enhanced Multifrequency Signaling for Feature Group D \(E911\), page 190](#)
- [Call Progress Analysis, page 191](#)
- [T.38 Fax Relay Statistics, page 192](#)

## SIP Support for Media Forking

The Session Initiation Protocol (SIP) Support for Media Forking feature provides the ability to create midcall multiple streams (or branches) of audio associated with a single call and then send those streams of data to different destinations. This feature allows service providers to use technologies such as speech recognition, voice authentication, speech recording, and text-to-speech conversion to provide sophisticated services to their end-user customers. An example is a web-browsing application that uses voice recognition and text-to-speech (TTS) technology to make reservations, verify shipments, or order products.

To enable the media forking feature on the Cisco AS5350XM, Cisco AS5400XM, and Cisco VGD 1T3 voice gateways, enter the **media forking** command in dial-peer configuration mode, voice-service voip configuration mode or global configuration mode:

```
dial-peer voice xxx voip
media forking
or
voice-service voip
media forking
or
voice class media tag
```

For more information, refer to the “[SIP Support for Media Forking](#)” section in “[Configuring SIP Connection-Oriented Media Forking and MLPP Features](#)” in the *Cisco IOS SIP Configuration Guide*.

## Automatic Switch to G.Clear Codec when ISDN Bearer Capability IE Indicates 64K Unrestricted Bearer

Switching automatically to the G.Clear codec when the ISDN bearer capability information element (IE) indicates 64K unrestricted digital information is required to ensure consistency between Cisco AS5350XM, Cisco AS5400XM, and Cisco VGD 1T3 voice gateways and Cisco integrated services router (ISR) voice gateways. Without this feature, the gateway rejects a call with 64K unrestricted bearer capability on a voice interface because it is considered a data call, not a voice call.

There are no modifications to the command-line interface for this feature. The capability is added by default when you are configuring voice ports on the gateway platforms.

## Private Line Automatic Ringdown for Trading Turrets Connection

The Private Line Automatic Ringdown for Trading Turrets feature improves connection service for turrets in the financial industry—primarily for corporations and enterprises that use turrets and POTS telephones for trading. Implementation of this feature ensures that a call between traders on a PLAR connection will be maintained if one of the traders goes on-hook or on-hold. This new capability also ensures that bandwidth is used only when needed.

For more information about this feature, refer to the [Private Line Automatic Ringdown for Trading Turrets](#) document.

## Connection Trunk and Transparent CCS for Trading Turrets

The Connection Trunk feature creates a permanent VoIP call between two endpoints. This permanent call allows the connection trunk to pass a supplemental signal, such as hookflash or point-to-point hoot-n-holler, between the two endpoints.

This feature sets up permanent trunks, bridging PBX networks over a WAN cloud. The gateways transparently switch the data to the other end of the trunk without interpreting the data in the bearer or signaling channels. Each DS0 under the controllers must be configured either to one of the channel-associated signaling (CAS) variants or to ext-sig for T-CCS. The voice ports have to be configured with the correct connection number and the dial peers have to be configured to direct the setup message that is generated by the router internally to the other end.

For more information about this feature, refer to the [Private Line Automatic Ringdown for Trading Turrets](#) feature guide.

## Modem Pass-through Using SIP re-invite

The Modem Pass-through Using SIP re-invite feature brings modem pass-through into compliance with Session Initiation Protocol (SIP) standards. Upon detection of a modem tone, a SIP re-invite is sent in order to adjust the speed to G.711 codec mode on both gateways (instead of sending a Named Signaling Event (NSE) across to the peer gateway). This enables the Cisco AS5350XM, Cisco AS5400XM, and Cisco VGD 1T3 voice gateways to interoperate with other gateways. Upon the occurrence of the phase reversal, an NTE packet is sent to the peer to ensure that echo cancellation (ECAN) is disabled.

As part of the re-invite, the gateway also sends an additional attribute to indicate that the call is modem pass-through. If the call is a fax pass-through, the additional attribute is not sent. Thereafter, upon detection of a phase reversal, a Named Telephone Event (NTE) packet is sent across to the peer gateway to indicate the occurrence of phase reversal. The detection of phase reversal initiates disabling echo cancellation. On the peer side, ECAN is disabled after the NTE packet is detected. If the terminating gateway is a Cisco gateway, then all of the re-invite would function as described. If the terminating gateway is not a Cisco gateway, the re-invite may not indicate that it is a modem. In this case, the originating gateway assumes that it is a fax pass-through call.

Figure 1 shows a topology and sequence for modem pass-through call flow through a protocol.

**Figure 1**      **Modem Pass-through Call Flow through a Protocol**



The following steps describe the progress of a call based on the initial recognition of a 2100-Hz tone ( $\pm 15$  Hz) (as shown in [Figure 1](#)):

1. Switch the active codec in use on the call to G.711 (if a codec other than G.711 was previously in use).
2. Disable the high pass filter.
3. Disable voice activity detection (VAD) and comfort noise generation (CNG).
4. Switch from any adaptive or dynamic jitter buffer in use to a fixed-length jitter buffer. (A depth of 200 ms is recommended when switching to a fixed-length jitter buffer.)

To enable the modem pass-through feature, use the **modem passthrough** command in voice service configuration mode:

```
modem passthrough {nse | protocol codec {g711alaw | g711ulaw [redundancy]}}
```

- Select the **nse** keyword to use NSEs
- Select the **protocol** keyword to use SIP/H.323 protocol to signal modem pass-through
- Select the **codec g711alaw** keyword to indicate G.711 a-law (64,000 bps) for E1
- Select the **codec g711ulaw** keyword to indicate G.711 mu-law (64,000 bps) for T1
- Select the **codec g711 ulaw redundancy** keywords to use packet redundancy for modem traffic (in compliance with RFC 2198)

For more information about modem pass-through, refer to the “  
*Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide*”

## Voice Port Bit Conditioning

**cas-custom**

**no**

You can configure the system to deviate from a country's default settings as defined by Cisco. To do this, choose **invert-abcd** as an optional keyword:

**cas-custom channel { define | ignore | invert-abcd }**

The **invert-abcd** keyword inverts the ABCD bits before TX and after RX. This feature is disabled by default, which is the ITU default.

## Voice Trunk Conditioning and Trunk Conditioning OOS

The voice trunk conditioning feature enables you to create a voice class, configure specific signaling attributes to the voice class, and then map the attributes in the voice class to either a Voice over Frame Relay, Voice over ATM, or a Voice over HDLC dial peer. Using the voice class, you can define the keepalive-signaling packet interval and the signal pattern (ABCD) bit pattern for Cisco-trunk (private-line) calls.

Trunk-conditioning signaling attributes apply to permanent point-to-point voice connections (private lines and tie lines) that you create using the **connection trunk** command.

Trunk conditioning enables control over Cisco private-line calls that are sent over Frame Relay or ATM networks. When private-line or tie-line calls are sent between two PBXs, fault indications are sent to the sending PBX. If the call fails, the PBX can select an alternate path to route the calls. Selecting an alternate path applies to analog connections or digital T1/E1 using channel-associated signaling (CAS) ABCD signaling. It does not apply to common-channel signaling (CCS).

When T1/E1 CAS is carried in transparent pass-through mode for arbitrary, unknown, or unsupported CAS protocols, you must define on-hook or idle patterns so that the digital signal processor (DSP) code can sense the idle call state and shut off the flow of voice packets when no active call is in progress. This mode provides an additional idle bandwidth-saving mechanism for those cases when voice activity detection (VAD) is not desired.

For more information about this feature, refer to the *Trunk Management Features* document.



### Note

An out-of-service (OOS) condition can be signaled using an ABCD bit pattern that is different from the busy or seized state. The difference enables the PBX to distinguish between OOS and congestion. For more information, refer to the *Trunk Management Features* document.

## Enhanced Multifrequency Signaling for Feature Group D (E911)

The Enhanced Multifrequency for Feature Group D and Analog CAMA Trunks feature enhances the 911 interconnect capabilities of Cisco IOS-based gateways. This document describes E911 support requirements, which include support for Enhanced Multifrequency signaling for Feature Group D and

Analog Centralized Automated Message Accounting (CAMA) signaling protocols per National Emergency Number Association standards. This feature supports 20-digit Automatic Number Identification (ANI) requirements and mapping of remote party IDs (RPID) to PANI.

For more information, refer to the [Enhanced MF for FGD and Analog CAMA Trunks](#) document.

## Call Progress Analysis

To enable answering machine detection for contact center applications, an enhanced call progress analysis (CPA) function for the Outbound Option Dialer is implemented in Cisco IOS Release 12.4(24)T. The answer machine and answer machine terminating tone detection algorithm is ported from the IPCC Dialer into the PVDM2 DSP voice feature card (VFC) and ISR PVDM2 DSP. The fax/modem detection portion uses existing logic in the PVDM2 DSP.

This feature is enabled by the following Cisco IOS commands in voice service voip configuration mode:

- `[ configuration ]`

The basic command enables the call progress analysis algorithm for all outbound VoIP calls in the voice gateway. If you select one of the optional keywords, you can specify other parameters for CPA threshold values and CPA timing values.

- `cpa` :

– `cpa threshold noise-level max {9db 12db 15db 18db 21db}`

– `cpa threshold noise-level min {-45dBm0 -50dBm0 -55dBm0 -60dBm0}`

value is –50 dBm0.

`cpa threshold noise-level min {-55dBm0 | -60dBm0 | -65dBm0 | -70dBm0}`

This command sets the minimum threshold that the CPA algorithm uses to measure the noise floor level. If the measured noise floor level is less than the value specified in this command, the measured noise floor level is set to the configured minimum noise floor level. The default value is –60 dBm0.

- `cpa timing` (in milliseconds):

`cpa timing live-person 1 ... 60000`

This command sets the timing that the CPA algorithm uses to determine whether or not a call is answered by a live person. The default value is 2500 ms.

`cpa timing noise-period 1 ... 60000`

This command sets the maximum time that the CPA algorithm uses to measure the noise floor level at the beginning of a call. The default value is 100 ms.

`cpa timing silent 1 ... 60000`

This command sets the minimum silent duration after active speech is detected in order for the CPA algorithm to determine that a call is answered by a live person. The default value is 375 ms.

`cpa timing term-tone 1 ... 60000`

This command sets the maximum time that the CPA algorithm waits for the answering machine terminating tone after the answering machine is detected. The default value is 15000 ms.

— ...

This command sets the maximum time that the CPA algorithm should wait before timing out if it doesn't detect a voice signal. The default value is 3000 ms.

— ...

This command sets the minimum voice duration for the CPA algorithm to consider it a valid speech signal. The default value is 112 ms.

For more information about the use of these commands, refer to the [Cisco IOS Voice Command Reference](#).

## T.38 Fax Relay Statistics

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# Additional References

## Related Documents

Related Topic	Document Title

## Standards

Standard	Title

## MIBs

MIB	MIBs Link
•	

## RFCs

RFC	Title

## Technical Assistance

Description	Link
<p>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</p> <p>To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds.</p> <p>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</p>	<p><a href="http://www.cisco.com/techsupport">http://www.cisco.com/techsupport</a></p>

## Command Reference

[vr\\_book.html](#). For information about all Cisco IOS commands, use the Command Lookup Tool at <http://tools.cisco.com/Support/CLILookup> or the [http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all\\_book.html](http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all_book.html), at

- `{ | | }`
- `[ | ]`
- `{ | } { | [ ] }`

## Feature Information for Voice Enhancement Features for Cisco IOS Release 12.4(24)T

Table 1 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to <http://www.cisco.com/go/cfn>. An account on Cisco.com is not required.



### Note

Table 1 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

**Table 1**      **Feature Information for Voice Enhancement Features for Cisco IOS Release 12.4(24)T**

Feature Name	Releases	Feature Information

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