# Single and High-Density Voice over IP Support for the Cisco AS5300/Voice Gateway

## Feature Overview

Cisco IOS Release 12.0(2)XH was the first (early deployment) release to support single and high-density VoIP support for the Cisco AS5300 (96/120 calls per system). Cisco IOS Release 12.0(4)XH1 (this release) adds support for the following features:

- H.323 Version 2
- H.323-based alternate Gatekeeper function)
- Debit Card Accounting/Remote Authentication Dial-In User Service (RADIUS) attributes
- Open Settlements Protocol
- Tool Command Language (TCL)-based Interactive Voice Response (IVR)
- Time Division Multiplexing (TDM) hairpinning
- Support for channelized T1 (CT1+) and channelized E1 (CE1+) serial port feature cards or eight T1/E1 feature cards for Frame Relay, High-Level Data Link Control (HDLC), and Point-to-Point Protocol (PPP) encapsulation and backhaul of Voice over IP (VoIP) traffic
- Feature Group D passing or overwriting of Answer Number Identification (ANI) digits on outbound calls

For complete information about Cisco IOS Release 12.0 XH, see *Release Notes for Cisco AS5300 Universal Access Server/Voice Gateway for Cisco IOS Release 12.0 XH/XH1*.

## New with VCWare

Cisco IOS 12.0(4)XH1 and VCWare Version 4.10 are the first releases that introduce feature parity (except for density) between the earlier generation AS53-6VOX (TI-C542-based) systems and the latest AS53-VOXD (TI-C549-based) high-density systems: There are now C542 and C549 versions of VCWare 4.10. For more information on Cisco IOS and VCWare releases, see "Cisco IOS Release and VCWare Compatibility."

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**Note** If you are running Cisco IOS Release 12.0(4)XH or 12.0(5)T2 with c542 feature cards on an earlier generation system, you must run the C542 version of Cisco VCWare Version 4.04.

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The following features are supported with VCWare Version 4.10:

- Three CODEC feature sets
  - "All CODECs" (G.711, G.729, G.723.1, G.726, and G.728) and Fax Relay for the high-density AS53-VOXD, TI c549-based feature cards only.
  - "All CODECs" and Fax Relay supporting early generation Cisco AS53-6VOX, TI c542-based feature cards only. Use this feature set for early generation voice feature cards and for upgrading earlier generation field units to provide feature compatibility with the latest high-density voice card feature set in VC-SWA-4.10.
  - "Limited CODECs" feature subset supporting G.711 and G.729(a) CODECs, and Fax Relay for the high-density AS53-VOXD, TI c549-based feature cards. This is only available for the VOXQ-based feature cards and voice bundles, which are low-cost alternatives to the "All CODEC" versions. However, VOXQ bundles are only available with high-density AS53-VOXD, TI c549-based feature cards.

**Note** The "Limited CODECs" feature set uses the G.729(a) CODEC. The "All CODECs" feature set uses the G.729 CODEC. The G.729(a) CODEC is a less complex version of G.729 CODEC. However, both CODECs are fully interoperable.

- CODEC negotiation: G.711 a-law, G.711 u-law, G.723.1, G.723.1 Annex A, G.729, and G.729 Annex B
- Support for G.726 and G.728
- G.729 "pre-IETF" supported as new CLI variant
- 14.4kb/s Fax Relay with 12.0(4)XH1
- Dual Tone Multi-Frequency (DTMF) Out-of-Band Relay through standards-based H.245 or Cisco RTP (RealTime Transport Protocol)
- Configurable packet/frame size

For more information about CODEC feature sets, see "New and Changed Information" in *Release Notes for Cisco VCWare Version 4.10 for Cisco AS5300/Voice Gateway* (78-6309-01).

## Frame Relay/HDLC/PPP Encapsulation and Backhaul of VoIP Traffic

12.0(4)XH1 is the first Cisco AS5300/Voice Gateway IOS release that supports the use of serial ports or T1/E1 ports for Frame Relay/HDLC/PPP encapsulation and backhaul of VoIP traffic: All previous Cisco IOS releases limited the Cisco AS5300/Voice Gateway to passing calls between TDM (T1/E1) ports and Ethernet (10/100 Base T) ports and only supported the use of the earlier generation 4CT1/4CE1 feature cards. Encapsulating and backhauling VoIP traffic onto Frame Relay/HDLC/PPP links required a separate router on the same Ethernet network to encapsulate and pass VoIP traffic into a serial backhaul link (and vice versa).

Cisco IOS Release 12.0(4)XH1 supports the use of the Cisco AS5300 4CT1+ serial port and 4CE1+ serial port feature cards, previously only supported for AS5300 access server (modem/ISDN termination) applications.

**Note** Cisco recommends that you use dedicated routers, such as Cisco 3600, 7200, and 7500 to backhaul VoIP traffic over Frame Relay/HDLC/PPP links.

Furthermore, if you use AS5300/Voice Gateway ports for Frame Relay, HDLC, or PPP encapsulation, Cisco recommends that you only support voice traffic. IP Precedence and Weighted Fair Queueing mechanisms may not guarantee acceptable voice quality and low latency in mixed voice and data environments. For example, FRF.12 is not supported on the AS5300/Voice Gateway in this release.

# H.323 Version 2 Support

H.323 Version 2 Support upgrades Cisco IOS software to comply with the mandatory requirements in the Version 2 specification. This upgrade enhances the existing Voice Over IP GateWay, the Multimedia Conference Manager (GateKeeper and Proxy), and the dual tone multifrequency (DTMF) digital relay using H.245.

H.323 Version 2 defines a lightweight registration procedure that requires full registration for initial registration, but uses an abbreviated renewal procedure to update the gatekeeper and minimize overhead. Lightweight registration requires each endpoint to specify a TimeToLive (TTL) value in its Registration Request (RRQ) message.

The H.323 Version 2 gateway supports the registration of fully-qualified E.164 numbers with the gatekeeper for phones connected directly to the gateway. Tunneling through H.225 User-to-User Information Element (UUIE) facilitates transparent handling of supplementary services between two endpoints through a VoIP network. This eliminates the need to interpret various supplementary signaling messages in the VoIP gateways.

For more information, see "Gateway Support for Alternate Gatekeeper."

# Single Density Voice Support with DSPM-542

This feature implements voice support on the Cisco AS5300 by using DSPM-542 digital signal processor (DSP) modules.

The benefits of voice features include support for:

- Coder Negotiation
- G.723.1 and G.729 voice coders
- 14.4kb/s FAX Relay
- DTMF Digit Relay through RTP
- CODEC negotiation

This release supports a C542-based VCWare that provides CODEC and feature interoperability between the earlier generation, TI-C542-based AS5300/Voice Gateways, and the latest high-density versions. This release supports parallel C542-based VCWare/DSPWare and C549-based VCWare/DSPWare.

**Note** The C542-based VCWare does not increase the number of calls supported on earlier generation voice feature cards. Increasing support to 96/120 channels requires the latest generation (C549-based, AS53-VOXD-based) voice feature cards.

# High-Density Voice Support with DSPM-549

This release implements high-density voice support on the Cisco AS5300 by using DSPM-549 digital signal processor (DSP) modules. When equipped with Voice Feature Cards (VFCs) and voice-enabled Cisco IOS software, the AS5300/Voice Gateway supports carrier-class VoIP and FAX over IP services.

High-density voice support increases the voice capacity of a Cisco AS5300 up to 120 channels. This increase in voice support provides the voice density of up to four T1 lines (96 voice or FAX calls) or four E1 lines (120 voice or FAX calls).

A fully configured AS5300/Voice Gateway can support up to two high-density (48/60 channel) voice feature cards. Therefore, the system supports up to 96/120 simultaneous voice/fax calls (4T1/E1 density).

The benefits of high-density voice features include:

- Low cost per voice channel
- Support for industry-standard voice CODECs, including G.711, G.729, and G.723.1
- Support for out-of-band dual-tone modulation frequency (DTMF) transport for coders that do not optimally transport DTMF
- Support for CODEC negotiation
- Configurable voice packet sizes

# Open Settlements Protocol For Packet Voice

Internet voice telephony is often used for toll bypass by using an existing data network or the Internet instead of PSTN trunking. Calls of this nature require an originating and terminating gateway to be completed. When the originating and terminating voice gateways are owned by two different carriers, settlement between these carriers is required. The Open Settlements Protocol for Packet Voice project implements a standardized settlement protocol which can be implemented between different vendors gateways and voice settlement servers.

The Cisco gateway-based Open Settlements Protocol interacts between carriers to create a single authentication at initialization. The authentication is the basis for the establishment of a secure communication channel between the Settlement system and the infrastructure component.

**Note** Secure Socket Layer (SSL) is not enabled for this release of Open Settlements Protocol.

This channel then allows the following three types of transactions to be handled:

- Call routing—The Settlement system can either accept a gateway endpoint from the requestor or assign one for the requestor.
- Call authorization—Based on the terminating endpoint address, the Settlement system determines whether the requesting gateway is permitted to originate calls for the terminating gateway. If the call is authorized, the Settlement system generates a token that allows the terminating gateway to accept the call.
- Call detail reporting—Each endpoint in a call leg reports when the call stops with the usual call details. The Settlement system reconciles the different reports of the calling and called parties and generates billing information. Call details are reported on a call-by-call basis.

**Note** Before downloading the software images containing Open Settlement Protocol for Packet Voice, see "Encryption Limitation on Open Settlements Protocol for Packet Voice" on page 7.

For more information on the Open Settlement Protocol, see "*Settlement for Packet Telephony on Cisco Access Platform*."

## Debit Card Billing/RADIUS/Vendor Specific Attribute Enhancements for Packet Telephony

The Debit Card feature offers service providers debit accounting for calling services. The Debit Card feature and RADIUS-specific enhancements also support Vendor Specific Attributes (VSA). The Debit Card for Packet Telephony on Cisco AS5300 works in tandem with the Cisco Interactive Voice Response (IVR) feature. The IVR voice scripts have been modified to use Tool Command Language (TCL) scripts.

The feature components consist of IVR functionality in Cisco IOS software that works in connection with an integrated third-party billing system. This includes the ability to maintain per-user credit balance information through a RADIUS interface to the Cisco IOS software. When these features are implemented, the billing system and IOS software functions enable a carrier to authorize voice calls and to debit individual user accounts in real time at the edges of a VoIP network without requiring external service nodes.

For more information on Debit Card Billing, see "*Debit Card for Packet Telephony on Cisco Access Platforms*."

## Interactive TCL-based Voice Response

Cisco is building voice gateways to connect more traditional telephone networks to voice over IP (VoIP) networks. Customers who are installing VoIP networks often need a mechanism at the gateway to present a customized interface to the caller. The interactive voice response (IVR) feature was first made available to customers with Cisco IOS Release 11.(3)NA2 with the Service Provider VoIP feature set. IVR, with the addition of scripts using Tool Command Language (TCL), is being introduced with Cisco IOS Release 12.0(4)XH/XH1, which is compatible with TCLware 1.0.x.

**Note** These TCL IVR scripts are the default scripts that must be used with the IVR application in Cisco IOS Release 12.0(4)XH/XH1 and later releases.

IVR consists of simple voice prompting and digit collection to gather caller information for authenticating the user and identifying the destination. IVR provides the ability to:

- Play customized prompts
- Collect account numbers and PINs
- Collect destination phone numbers
- Perform AAA tasks interacting with a variety of servers

**Note** Most changes by Cisco in the TCL-based IVR scripts will not require IOS revision updates. Rather, updates will be downloadable through Flash—unlike the previous IVR implementation where scripts were imbedded in IOS. This will add more flexibility in subsequent scripts.

For more information on TCL scripts, see "*Debit Card for Packet Telephony on Cisco Access Platforms*." For more information on IVR, see "*Configuring Interactive Voice Response for Cisco Access Platforms*."

## H.323 Alternate Gatekeeper

This release adds support for the H.323 "alternate gatekeeper" field on the AS5300/Voice Gateway. This field was added to provide future support for specific third-party gatekeepers. Cisco gatekeepers do not support this field.

# Benefits

The benefits of these high-density voice feature enhancements include:

- Support for H.323 Version 2
- Expanded support for industry-standard voice CODECs, including G.726, G.728, and G.729a
- Support for Frame Relay/HDLC/PPP encapsulation by using T1/E1 ports and serial backhaul ports on Quad+ T1/E1 cards
- Support for TDM hairpinning without requiring the sustained use of digital signal processor modules (DSPs)
- Support for out-of-band dual tone multi-frequency (DTMF) digit relay

# Restrictions

The following general restrictions apply to this release:

- High-density voice cards and DSP modules (c549-based) cannot be in the same system as c542-based voice cards.
- If G.711 is used for more than 100 calls, you must enable voice activity detection (VAD).
- Cisco AS5300/Voice Gateway is rated to support up to two calls per second on a sustained basis. The AS5300/Voice Gateway can process higher call rates, but the call success rate and CPU utilization can be adversely affected.

## Restrictions for Backhaul/Encapsulation

Encapsulating and backhauling VoIP traffic onto Frame Relay/HDLC/PPP links requires the use of dedicated routers (such as Cisco 3600, 7200, 7500) on the same Ethernet network to encapsulate and pass VoIP traffic into a serial backhaul link (and vice versa).

Voice quality is directly affected by latency, which affects the CPU load. The following section highlights some key requirements and limitations associated with use of native FR/HDLC/PPP encapsulation for voice applications by using the Cisco AS5300/Voice Gateway.

To maintain high performance—including low latency (less than 200 ms round trip delay)—high-call success rate (99% CSR), and optimal voice quality, Cisco supports the use of native FR/HDLC/PPP encapsulation with the following restrictions:

- Enable VAD and voice compression CODECs for all calls (for example, G.729, G.723.1)
- Do not exceed a sustained call rate of 1 call per second.
- Enable only two serial ports. Limit each port to a bandwidth/clock rate of no more than 2Mbs. Or, limit one T1/E1 channelized backhaul port to between 1-3MBs bandwidth.
- Maintain a 2:1 ratio of TDM calls/bandwidth to encapsulated VoIP bandwidth (for example, 48/60 TDM calls encapsulated and carried through a 1 - 2MB/s serial link or into a 1 T1/E1/PRI link). Route four T1/E1 of TDM traffic (96/120 calls) through two 2Mbs serial backhaul links.
- Disable Compressed Real Time Protocol (CRTP). This is the default.

**Caution** Do **not** enable CRTP on the AS5300/Voice Gateway, except for a few very limited cases, specifically where less than 10 - 24 total calls will be processed. Offload Cisco routers (for example 7200/7500 series) dedicated for any CRTP requirements. CRTP is processed switched and creates a significant load on the main AS5300 CPU/router resource.

Even in the case of very low-expected call volumes, first test a simulation of the proposed network environment to validate achievable call success rate and low latency with CRTP enabled. If you need native CRTP, limit the call rate to 1 call every 2 seconds.

If you do use CRTP for more than 10 - 24 simultaneous calls, it will have a negative effect on the sustained call success rate, packet latency, and voice quality of the AS5300.

These caveats are provided for guidance, and there are a wide variety of other backhaul scenarios that can be supported by the AS5300/Voice Gateway, particularly for low-volume applications. When considering other configurations, consider that there are many other VoIP features, processes, and conditions that contribute to system load, including:

- Call volume
- H.323 RAS transactions
- AAA/RADIUS negotiation
- SNMP polling
- IVR
- Open Settlements Protocol transactions
- Enabling debugs (including “show” commands)

# Encryption Limitation on Open Settlements Protocol for Packet Voice

Open Settlements Protocol for Packet Voice is offered only in crypto images; they are under export controls. All users must be entitled before they can receive 56 or 56i images. See “Related Documents” for information on applying for entitlement.

Once you are entitled, you can see crypto images in the upgrade planner and you do not have to entitle yourself again—unless you come from a different host. You do not have to entitle yourself for every release because entitlement is good for all releases.

# Related Documents

For related information on this feature, see the following documents:

- *Release Notes for Cisco AS5300 Universal Access Server/Voice Gateway for Cisco IOS Release 12.0 XH* (78-6618-02)
- *Release Notes for Cisco VCWare Version 4.10 for Cisco AS5300/Voice Gateway* (78-6309-01)
- *High-Density Voice over IP Support for the Cisco AS5300/Voice Gateway* for Cisco IOS release 12.0(2)XH

# Supported Platform

Cisco AS5300

# Supported MIBs and RFCs

## MIBs

- CISCO MIB
- DIAL-CONTROL-MIB.my
- CISCO-DIAL-CONTROL-MIB.my
- CISCO-VOICE-DIAL-CONTROL-MIB.my
- CISCO-VOICE-IF-MIB.my
- CISCO-DSP-MGMT-MIB.my

For descriptions of supported MIBs and how to use MIBs, see Cisco's MIB website on Cisco Connection Online (CCO) at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

## RFCs

None

# Prerequisites

The following Cisco IOS Release images are required for 549 DSP modules:

- Cisco IOS Release 12.0(4)XH1 or later releases for high-density VoIP features on the T1 549 DSP modules.
- VFC ROM monitor software Version 1.2 or 1.3 is compatible with your installed Cisco IOS image. VCWare Version 4.10 (or a later release) requires Cisco IOS Release 12.0(4)XH1 or later releases. For more information on VCWare and IOS release compatibility, see "Cisco IOS Release and VCWare Compatibility Matrix."

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**Note** In certain countries, use of these products or provision of voice telephony over the Internet may be prohibited or subject to laws, regulations, or licenses, including requirements applicable to the use of the products under telecommunications and other laws and regulations; customers must comply with all such applicable laws in the countries where customers intend to use the product.

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Before you can configure your Cisco AS5300 to use VoIP, you must first complete the tasks listed below.

**Step 1** Establish a working IP network. See Cisco’s *Network Protocols Configuration Guide, Part I*.

**Step 2** Complete basic configuration for the Cisco AS5300 (that is, configure a host name, password, Ethernet interface, and ISDN PRI lines). For more information about these configuration tasks, see *Cisco AS5300 Universal Access Server Software Configuration Guide*.

**Step 3** Install the VoIP card (VFC) into the appropriate slot of your Cisco AS5300 access server. Each VFC can hold up to five digital signal processor modules, enabling processing for up to 30 channels with single-density DSPs or 60 channels with high-density DSPs.

**Step 4** Complete your company's dial plan and establish a working telephony network based on your company's dial plan.

**Step 5** Integrate your dial plan and telephony network into your existing IP network topology. For detailed procedures, see *Cisco AS5300 Universal Access Server Software Configuration Guide*.

**Step 6** Configure Voice over IP.

# Cisco IOS Release and VCWare Compatibility

The Cisco VCWare compatibility matrix contains information about Cisco IOS and VCWare compatibility for the Cisco AS5300/Voice Gateway.

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**Note** VCWare is not backward compatible. Do not use it with Cisco IOS releases earlier than this release (12.0(4)XH1). Use each VCWare release with its compatible Cisco IOS release. VCWare Version 4.04 or a later release is required for use with Cisco IOS Release 12.0(2)XH, 12.0(2)XH1, and 12.0(5)T2. VCWare Version 4.10 or a later release is required with Cisco IOS Release 12.0(4)XH1.

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**Table 1 Cisco IOS Release and VCWare Compatibility Matrix**

| Cisco IOS Release | VCWare Version | DSPWare Version | New VCWare Feature |
|---|---|---|---|
| 12.0(4)XH1 | 4.10 or a later release | 3.1.10 or a later release | Support for high-density voice cards (c549-based DSP Modules, p/n AS53-VOXD) and earlier generation c542-based DSP modules. Also adds new CODECs, including G.726 and G.728.<br>**Note** VCWare 4.10 is released in 3 versions:<br>• c549-based "All CODECs" feature set, Cisco part number: VC-SWA-4.10. Supports G.711, G.729, G.726, G.723.1, G.728, and Fax Relay. File name: vcw-vfc-mz.c549.hc.4.10.bin<br>• c542-based "All CODECs" feature set, Cisco part number: VC-SWS-4.10. Supports G.711, G.729, G.726, G.723.1, G.728, and Fax Relay. File name: vcw-vfc-mz.c542.4.10.bin<br>• c549-based "Limited CODECs" feature set, Cisco part number: VC-SWQ-4.10. Supports G.711, G.729(a), and Fax Relay. File name: vcw-vfc-mz.c549.mc.4.10.bin |
| 12.0(4)XH<br>12.0(5)T2 (planned) | 4.04 or a later release | 3.1.7 or a later release | Support for c549-based DSP modules and c542-based DSP modules with two separate VCWare images. |
| 12.0(2)XH | 4.04 or a later release | 3.1.7 or a later release | Support for high-density voice cards (TI c549-based DSP Modules, p/n AS53-VOXD) and adds new CODECs, including G.723.1. |

# Configuration Tasks

Perform the following tasks to configure high-density voice for the Cisco AS5300:

- Identifying Voice Feature Cards
- Replacing Firmware with VCWare in VCWare Mode
- Replacing Firmware with VCWare in ROM Monitor Mode
- Configuring Voice Preference Groups

## Identifying Voice Feature Cards

The following steps identify the voice cards in the system and determine whether the VFC is in VCWare mode or ROM monitor mode. The mode determines how you download software to the VFC. If VCWare is not loaded on the VFC, you can use the ROM monitor mode to download the VCWare. Otherwise, the VCWare mode is active.

| Step | Command | Purpose |
|---|---|---|
| **1** | `5300>` **`enable`**<br>`Password: <`**`password`**`>`<br>`5300#` | Enter enable mode (also called Privileged EXEC mode).<br>Enter the password.<br>You have entered enable mode when the prompt changes to *router#*. |
| **2** | `5300#` **`show vfc`** `<0-2>` **`technology`** | Determine the type of voice feature card (requires VCWare mode).<br>If this command returns c542 or does not work, go to Step 4. |
| **3** | `5300#` **`show vfc`** `<0-2>` **`version vcware`** | Determine the complexity mode. The first line of the return (cor-vfc) identifies the card as being in high-complexity mode if the return is in the form: cor-vfc-hc-1.x.x.x.bin.<br>If the return is in the form: cor-vfc-mc-l.x.x.x.bin, the card is in medium complexity mode. |
| **4** | `5300#` **`show vfc`** `<0-2>` **`board`** | Determine the number of voice cards in the system, the slot number for each card, and the VFC mode (VCWare or ROM monitor) in which you are running.<br>The VFC mode is indicated as follows:<br>• VCWARE running for VCWare mode<br>or<br>• ROMMON for ROM monitor mode<br>Note the location and the mode type for each voice card. You will need this information when you upgrade the VCWare. |

Go to one of the following sections to continue the procedure to upgrade the VCWare:

- “Replacing Firmware with VCWare in VCWare Mode” if your voice card is running in VCWare mode

  or

- “Replacing Firmware with VCWare in ROM Monitor Mode” if your voice card is running in ROM monitor mode.

# Replacing Firmware with VCWare in VCWare Mode

Use these steps to download new voice software if your voice card is running in VCWare mode. When downloading from a diskette, first copy the software from the diskette to a TFTP server.

**Caution** Erasing the VFC files can result in system outage or the corruption of your VFC board. Ensure that you have the correct software version on your TFTP server.

| Step | Command | Purpose |
|---|---|---|
| **1** | 5300# **erase vfc** *slot_number*<br>This will erase the contents of VFC Flash.<br>Continue ?[y/n]: **y**<br><br>This will take some time. Please, wait...vfc | See the Caution that precedes this table.<br>Erase the contents of the VFC Flash memory in the selected voice card. You must erase the previous files to load a new image.<br>If this command fails, go to “Replacing Firmware with VCWare in ROM Monitor Mode." |
| **2** | 5300# **show vfc** *slot_number* **directory** | (Optional) Verify that the VFC Flash memory is empty. |
| **3** | 5300# **copy tftp: vfc:**<br>Address or name of remote host []?<br>**209.165.200.254**<br>Source file name []? /path/*filename*<br>Destination file name []? *filename*<br>Accessing tftp://*ipaddress*/path/*filename*...<br>Loading *filename* from *ipaddress* (via Ethernet0):<br>!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!![OK -<br>*size* bytes]<br><br>or<br><br>5300# **copy tftp vfc**<br>Voice card slot number <slot ? 1><br>Address or name of remote host [UNKNOWN]? *ipaddress*<br>Source file name? /path/*filename*<br>Destination file name [*filename*]? *filename*<br>! note, the destination filename is *IMPORTANT*<br>Accessing file '*filename*' on<br>*ipaddress*...<br>Loading *filename* from *ipaddress* (via Ethernet0):<br>!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!![OK -<br>*size* bytes] | Use the first copy command with Cisco IOS Release Versions 12.0(2)XH, 12.0(4)XH, 12.0(4)XH1, and 12.0(5)T2. Use the second copy command (no colons) with Cisco IOS Release Version 11.3(2)NA or later releases.<br>Use TFTP to download the new VCWare to VFC Flash memory. In this example, slot 1 is the default.<br>Note: Begin the destination image name with “vcw-” (all lowercase). Do not use any other character in front of vcw-; therefore, do not include a directory path at the beginning of the image name. Make sure that you enter the name correctly. If one character or space is wrong, the file is not recognized and is not copied. |
| **4** | 5300# **show vfc** slot_number **directory**<br>5300# | Verify that VCWare is in the VFC Flash memory by providing a list of files. |
| **5** | 5300# **clear vfc** *slot_number*<br>5300# | Reload the VFC code from Flash memory for the specified slot before you unbundle the new VFC image. |

| Step | Command | Purpose |
|---|---|---|
| 6 | 5300# **unbundle vfc** *slot_number* | Unbundle the DSPWare from the VCWare and configure the default file and capability lists with default values. This rewrites the default-file and capabilities lists. (See Steps 8 and 9.) |
| | Do you want to continue ? [y/n]: **y** | Press **y** when the prompt appears. |
| 7 | 5300# **show vfc** *slot_number* **directory**<br>5300# | Verify that the DSPWare is unbundled by providing a list of files. |
| 8 | 5300# **show vfc** *slot_number* **default-file**<br>5300# | Verify that the default file list is initialized by providing a list of files. |
| 9 | 5300# **show vfc** *slot_number* **cap-list**<br>5300# | Verify that the capability list is populated by providing a list of files. |
| 10 | 5300# **reload** | Reboot the Cisco AS5300 so these changes take effect. |

# Replacing Firmware with VCWare in ROM Monitor Mode

Use these steps to download new voice software if your voice card is running in ROM monitor mode. When downloading from a diskette, first copy the software from the diskette to a TFTP server, then, follow these steps.

**Caution** Erasing the VFC files can result in system outage or the corruption of your VFC board. Ensure that you have the correct software version on your TFTP server before continuing. For more information on software versions, see “Cisco IOS Release and VCWare Compatibility.”

| Step | Command | Purpose |
|---|---|---|
| 1 | 5300# **clear vfc** *slot_number* **purge**<br>debug vfc <slot#> start<br>debug vfc <slot#> cons flash erase<br><br>This will erase the contents of VFC Flash.<br>Continue ?[y/n]: **y**<br><br>This will take some time. Please, wait...vfc | See the Caution that precedes this table.<br>(This step is optional.) Erase the contents of the VFC Flash memory in the selected voice card. You must erase the previous files to load a new image.<br>Press **y** to continue. |

| Step | Command | Purpose |
|---|---|---|
| **2** | 5300# **copy tftp: vfc:**<br>Address or name of remote host []? *ipaddress*<br>Source file name []? /path/*filename*<br>Destination file name []? **filename**<br>Accessing tftp://*ipaddress*/path/<br>*filename*...<br>Loading *filename* from *ipaddress*<br>(via Ethernet0):<br>!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!![OK -<br>size bytes]<br><br>or<br><br>5300# **copy tftp vfc**<br>Voice card slot number <slot ? 1><br>Address or name of remote host [UNKNOWN]? *ipaddress*<br>Source file name? /path/*filename*<br>Destination file name [*filename*]? *filename*<br>! note, the destination filename is *IMPORTANT*<br>Accessing file '*filename*' on *ipaddress*<br>Loading *filename* from *ipaddress* (via Ethernet0):<br>!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!![OK -<br>size bytes] | Use the first copy command with Cisco IOS Release Versions 12.0(2)XH, 12.0(4)XH, 12.0(4)XH1, and 12.0(5)T2. Use the second copy command (no colons) with Cisco IOS Release Version 11.3(2)NA or later releases.<br>Use TFTP to download the new VCWare to VFC Flash memory. In this example, slot 1 is the default.<br>Note: Begin the destination image name with “vcw-” (all lowercase). Do not use any other character in front of vcw-; therefore, do not include a directory path at the beginning of the image name. Make sure that you write the name correctly. If one character or space is wrong, the file is not recognized and is not copied. |
| **3** | 5300# **show vfc** slot_number **technology**<br>5300# | Report the type of DSPs supported (C542 or C549). |
| **4** | 5300# **clear vfc** *slot_number*<br>5300#<br><br>Do you really want to reset this card ?<br>[y/n] **y**<br>Please, wait... | Reload the VFC code from Flash memory for the specified slot before you unbundle the new VFC image.<br><br>Press **y** to verify the reset function. |
| **5** | 5300# **unbundle vfc** *slot_number*<br><br>Do you want to continue ? [y/n]: **y** | Unbundle the DSPWare from the VCWare and configure the default file and capability lists with default values. This rewrites the default-file and capabilities lists. (See Steps 7 and 8.)<br><br>Press **y** when the prompt appears. |
| **6** | 5300# **show vfc** *slot_number* **directory**<br>5300# | Verify that the DSPWare is unbundled by providing a list of files. |
| **7** | 5300# **show vfc** *slot_number* **default-file**<br>5300# | Verify that the default file list is initialized by providing a list of files. |
| **8** | 5300# **show vfc** *slot_number* **cap-list**<br>5300# | Verify that the capability list is populated by providing a list of files. |
| **9** | 5300# **reload** | Reboot the Cisco AS5300 so these changes take effect. |

# Verifying VCWare or ROM Monitor Firmware Replacement

To check that you have successfully downloaded the software:

**Step 1** Enter the **show vfc** *slot_number* **directory** command to verify that the VCWare is in the Flash memory. Only one filename appears. If this command times out, start over with “Identifying Voice Feature Cards.”

```
5300# show vfc 1 dir

Files in slot 1 VFC flash:
        File Name              Size (Bytes)
1 .  vcw-vfc-mz.c549.3.0.bin    466168
2 .  btl-vfc-l.3.0.bin          4174
3 .  btj-vfc-l.3.0.bin          3868
4 .  jbc-vfc-l.3.0.bin          12756
5 .  cor-vfc-hc-l.3.0.bin       47754
6 .  bas-vfc-hc-l.3.0.bin       38634
7 .  fax-vfc-hc-l.3.0.bin       80794
8 .  cdc-g711-hc-l.3.0.bin      218
9 .  cdc-g726-hc-l.3.0.bin      218
10.  cdc-g729-hc-l.3.0.bin      31516
11.  cdc-g728-hc-l.3.0.bin      24334
12.  cdc-g723.1-hc-l.3.0.bin    47234
```

**Step 2** Enter the **show vfc** *slot_number* **default-file** and **show vfc** *slot_number* **cap-list** commands to verify that the DSPWare is unbundled and the default file list and cap-list are initialized.

```
5300# show vfc 1 def (full word is default-file)

Default File List for VFC in slot 1:

1. btl-vfc-l.3.0.bin
2. cor-vfc-hc-l.3.0.bin
3. bas-vfc-hc-l.3.0.bin
4. cdc-g729-hc-l.3.0.bin
5. fax-vfc-hc-l.3.0.bin
6. jbc-vfc-l.3.0.bin
7. btj-vfc-1.3.0.bin

5300# show vfc 1 cap-list

Capability List for VFC in slot 1:

1. cor-vfc-hc-l.3.0.bin
2. bas-vfc-hc-l.3.0.bin
3. fax-vfc-hc-l.3.0.bin
4. cdc-g711-hc-1.3.0.bin
5. cdc-g726-hc-1.3.0.bin
6. cdc-g729-hc.1.3.0.bin
7. cdc-g728-hc-1.3.0.bin
8. cdc-g723.1-hc-1.3.0.bin

5300#
```

# Configuring Voice Preference Groups

To create preferences lists that can be applied to any dial peer:

**Step 1** In configuration mode, enter the **voice class codec** *tag* command, where *tag* is the ID number you assign to this voice class preference list.

**Step 2** In configuration class mode, enter each preference ID number (from 1 to 10,000) followed by the CODEC type.

# Configuration, Verification, and Troubleshooting Tips

If you have trouble downloading the voice card software in VCWare mode, see the following suggestions:

**Step 1** Enter the **show vfc** *slot_number* **board** command to verify that the voice card is back up in VCWare mode.

```
5300# show vfc 1 board
VFC board state is UP, vfc status VCWARE running(0x4)
VFC board in slot 1 with 24 dsps
5300#
```

**Step 2** Determine if the VFC ROM version you are running is Version 1.1 or a later version.

```
5300# show vfc 1 ver vcw

Voice Feature Card in Slot 1:
     VCware Version      : 3.0
     ROM Monitor Version : 1.2
     DSPware Version     : 3.0
     Technology          : C549
```

**Step 3** Enter the **reload** command to reboot the Cisco AS5300 so these changes take effect.

If the system crashes after the new VFC cards have been placed in the system, it could mean that one or both of the VFC cards are not seated properly. Pull each VFC card partially out of its slot and reseat it firmly.

If a download failure occurs in VCWare mode and the VFC falls back to ROM monitor mode, see "Replacing Firmware with VCWare in ROM Monitor Mode."

Make sure that you count the number of DSP modules on both VFC cards to determine the total number of voice resources. Cisco supports up to five DSP modules per VFC. Then, enter the **show version** command to confirm that the total number of functioning voice resources equal the number of voice resources installed.

# Configuration Examples

This section provides the following configuration examples:

- Replacing VCWare Firmware
- Replacing ROM Monitor Firmware
- Creating and Applying a Voice Preference List

# Replacing VCWare Firmware

See the following example to download new voice software for a voice card running in VCWare mode:

```
5300# erase vfc 2
This will erase the contents of VFC Flash.
Continue ?[y/n]:yes
This will take some time. Please, wait...vfc
5300# show vfc 2 directory
5300# copy tftp: vfc:
Address or name of remote host []?
223.255.212.244
Source file name []? /path/vcw-vfc-mz.c549.3.0.bin
Destination file name []? vcw-vfc-mz.c549.3.0.bin
Accessing tftp://223.255.212.244/path/
vcw-vfc-mz.c549.3.0.bin...
Loading vcw-vfc-mz.c549.3.0.bin from 223.255.212.244
(via Ethernet0):
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
!![OK - 491256/982016 bytes]
5300# show vfc 2 directory
5300# clear vfc 2
5300# unbundle vfc 2


Do you want to continue ? [y/n]: y
5300# show vfc 2 directory
5300# show vfc 2 default-file
5300# show vfc 2 cap-list
5300# reload
```

# Replacing ROM Monitor Firmware

See the following example to download new voice software for a voice card running in ROM monitor mode:

```
5300# clear vfc 1 purge
debug vfc 1 start
debug vfc 1 cons flash erase

This will erase the contents of VFC Flash.
Continue ?[y/n]: y

This will take some time. Please, wait...vfc
5300# copy tftp: vfc:
Voice card slot number 1
Address or name of remote host [UNKNOWN]? 223.255.212.244
Source file name? /path/vcw-vfc-mz.c549.3.0.bin
Destination file name [vcw-vfc-mz.c549.3.0.bin]? vcw-vfc-mz.c549.3.0.bin
! note, the destination filename is *IMPORTANT*
Accessing file 'vcw-vfc-mz.c549.3.0.bin' on 223.255.212.244...
Loading vcw-vfc-mz.c549.3.0.bin from 223.255.212.244 (via Ethernet0):
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!![OK - 291292/291328 bytes]
5300# show vfc 1 technology
5300# clear vfc 1



Do you really want to reset this card ? [y/n] y
Please, wait...
5300# unbundle vfc 1




Do you want to continue ? [y/n]: y
5300# show vfc 1 directory
5300# show vfc 1 default-file
5300# show vfc 1 cap-list
5300# reload
```

# Creating and Applying a Voice Preference List

See the following example to create preference list 99 which can be applied to any dial peer:

```
5300(#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
5300(config)#voice class codec 99
5300(config-class)#codec preference 1 g711alaw
5300(config-class)#codec preference 2 g711ulaw bytes 80
5300(config-class)#codec preference 3 g723ar53
5300(config-class)#codec preference 4 g723ar63 bytes 144
5300(config-class)#codec preference 5 g723r53
5300(config-class)#codec preference 6 g723r63 bytes 120
5300(config-class)#codec preference 7 g726r16
5300(config-class)#codec preference 8 g726r24
5300(config-class)#codec preference 9 g726r32 bytes 80
5300(config-class)#codec preference 10 g728
5300(config-class)#codec preference 11 g729br8
5300(config-class)#codec preference 12 g729r8 bytes 50
5300(config-class)#end
#
5300(config)#
```

See the following example to apply preference list 99 to dial-peer 1919:

```
5300(#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
5300(config)#voice class codec 99
5300(config-class)#codec preference 1 g711alaw
5300(config-class)#codec preference 2 g711ulaw bytes 80
5300(config-class)#codec preference 3 g723ar53
5300(config-class)#codec preference 4 g723ar63 bytes 144
5300(config-class)#codec preference 5 g723r53
5300(config-class)#codec preference 6 g723r63 bytes 120
5300(config-class)#codec preference 7 g726r16
5300(config-class)#codec preference 8 g726r24
5300(config-class)#codec preference 9 g726r32 bytes 80
5300(config-class)#codec preference 10 g728
5300(config-class)#codec preference 11 g729br8
5300(config-class)#codec preference 12 g729r8 bytes 50
5300(config-class)#end
#
5300(config)#
#
(config)#dial-peer voice 1919 voip
(config-dial-peer)#voice-class codec 99
(config-dial-peer)#end
```

You can verify this action by entering the **show ru** command:

```
#show ru
dial-peer voice 1919 voip
voice-class codec 99
!
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw bytes 80
codec preference 3 g723ar53
codec preference 4 g723ar63 bytes 144
codec preference 5 g723r53
codec preference 6 g723r63 bytes 120
codec preference 7 g726r16
codec preference 8 g726r24
codec preference 9 g726r32 bytes 80
codec preference 10 g728
codec preference 11 g729br8
codec preference 12 g729r8 bytes 50
```

# Command Reference

This section documents the new **calling-number** command. All other commands used with this feature are documented in the Cisco IOS Release 12.0 command references and the online feature module: *High-Density Voice over IP Support for the Cisco AS5300/Voice Gateway* for Cisco IOS release 12.0(2)XH.

In Cisco IOS Release 12.0(1)T or later releases, you can search and filter the output for **show** and **more** commands. This functionality is useful when you need to sort through large amounts of output, or if you want to exclude output that you do not need to see.

To use this functionality, enter a **show** or **more** command followed by the "pipe" character (|), one of the keywords **begin**, **include**, or **exclude**, and an expression that you want to search or filter on:

*command* **|** {**begin** | **include** | **exclude**} *regular-expression*

See the following example of the **show atm vc** command where you want the command output to begin with the first line where the expression "PeakRate" appears:

**show atm vc | begin PeakRate**

For more information on the search and filter functionality, see the Cisco IOS Release 12.0(1)T feature module *CLI String Search*.

# calling-number

Enter the **calling-number** dial-peer configuration command to enable or disable automatic number identification (ANI) passing on outgoing calls for T1 CAS (channel associated signalling) configured interfaces. Enter the **no** form of this command to disable ANI configuration.

**calling-number** {**range** *string1 string2*} | {**sequence** *string1 string2 string3 string 4 string 5*} | {**null**}

**no calling-number**

## Syntax Description

| | |
|---|---|
| **range** | Specifies *string1* and *string2* as valid E164 telephone number strings. Both strings must be the same length (up to a maximum of 32 digits).<br><br>For strings longer than four digits, only the last four digits are used to specify the range and generate the ANI sequence by rotating through the range until reaching *string2*. The range then restarts at *string1* and repeats. If strings are less than four digits, then the entire string is used. |
| **sequence** | Configures a sequence of up to five discrete strings that are substituted for ANIs on successive calls by using the dial peer.<br><br>Strings must be valid E164 telephone number strings with a maximum length of 32 digits. |
| **null** | Suppresses the use of ANI. When you specify this option, no ANI is passed when you specify this dial peer. |

## Defaults

none

## Command Modes

Dial-peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(4)XH1 | This command was introduced. |

## Usage Guidelines

This command is designed for plain old telephone service (POTS) dial peers only.

You can use up to two information digits to prepend to the ANI string.

## Examples

See the following example to generate an ANI with the prefix 408555 and rotating through 1000 to 1005 for each call using this peer:

```
5300(config-dial-peer)# dial-peer voice 1 pots
5300(config-dial-peer)# calling-number range 4085551000 4085551005
```

See the following example to generate strings to be substituted for ANIs for the next five calls using this peer:

```
5300(config-dial-peer)# dial-peer voice 1 pots
5300(config-dial-peer)# calling-number sequence 1111 1112 1113 1114 1115
```

# List of Terms and Acronyms

**AAA**—Authentication, Authorization, and Accounting.

**ANI**—Answer Number Identification (calling party number).

**CLI**—Command Line Interface, IOS user command, and configuration system.

**CO**—Central Office.

**CRTP—**Compressed Real Time Protocol.

**D-channel**—Signaling channel; pathway for out-of-band call control.

**DMA**—Direct Memory Access (used by VC to transfer data to the AS5300 motherboard).

**DNIS**—Dialed Number Identification Service.

**DSP—**Digital Signal Processor.

**DTMF—**Dual Tone Multifrequency.

**E.164 Address—**In this context, the telephone number.

**EANA—**Exchange Access North American.

**FGD—**Feature Group D. Identifies a standardized service available to carriers delivered on a channelized T1 line.

**Gateway**—H.323 VoIP gateway is the point where a circuit-switched call is encoded and encapsulated into IP packets.

**GateKeeper—**In the context of this feature module, H.323 VoIP gatekeeper refers to the Cisco IOS-based gatekeeper.

**H.323 RAS**—Registration, Admission, and Status. The RAS signaling function performs registration, admissions, bandwidth changes, status, and disengagement procedures between the VoIP Gateway and Gatekeeper.

**Hairpin Re-routing**—also known as Hairpinning. Rerouting a call back to the PSTN at the originating VoIP gateway–usually because a VoIP route is unavailable or the PSTN is judged to be the most cost-effective alternative.

**HSRP—**Hot Standby Routing Protocol. Previously existing protocol for failover to standby gatekeeper/router in a redundant router configuration.

**ISAC** —ISDN Subscriber Access Controller.

**ISDN** —Integrated Services Digital Network.

**ITU** —International Telecommunication Union (formerly CCITT).

**IVR**—Interactive Voice Response. The system plays a message to the calling party and can also collect information from the calling party.

**LEC** —Local Exchange Carrier.

**On-net—**In this context of this feature module, on-net calls are voice over IP calls on the IP Network.

**Off-net**—In this feature module, off-net calls are those that are circuit-switched calls, whether they are switched by the PSTN or a PBX.

**PBX—**Private Branch eXchange.

**PRI** —ISDN Primary Rate Interface.

**PSTN** —Public Switched Telephone Network.

**Q.931**—ITU-T recommendation for Digital Subscriber Signalling System No. 1 Network Layer.

**RTP**—Real Time Protocol.

**SNMP—**Simple Network Management Protocol.

**TDM**—Time Division Multiplexing.

**VFC—**Voice Feature Card.

**VoIP—**Voice over IP.