Welcome to **Cisco DocWiki**. We encourage registered Cisco.com users to contribute to this wiki to improve Cisco product documentation. Note that you cannot log in to DocWiki with Cisco.com "guest" account credentials.

See Terms of Use and About DocWiki for more information about Cisco DocWiki.
Select the "Edit" tab to edit an article or select the "Leave a Comment" tab to submit questions or comments about the article.

Click here to return to the Cisco IOS documentation on www.cisco.com.
==Print multiple DocWiki pages at once as a **PDF** document. See the DocWiki Print Page.==

This article provides guidance for troubleshooting issues that may appear when using Cisco IOS software for voice applications. Tools and methodologies are described that help to recognize a problem, determine its cause, and find possible solutions.

Contents

Troubleshooting Cisco IOS Voice Overview

To troubleshoot problems with voice networks, you must follow the call both inside the router and outside on the network in order to isolate the problem. You must understand the relationship of dial peers and call legs to follow the calls. For detailed information on dial peers and call legs, refer to Dial Peer Configuration on Voice Gateway Routers.

The following articles contain information about call flow:

- Call Flow Overview
- Troubleshooting Tools

Debug Command Output on Cisco IOS Voice Gateways

The debugging capability for Cisco voice gateways enables you to identify and track a specific call in a multiple-call environment. This capability allows you to correlate call information between gateways or to identify specific debug messages associated with a single call when multiple voice calls were simultaneously active.

Voice debug output contains a standardized header to the debug outputs of multiple voice modules, such as voice telephony service provider (VTSP), call control application program interface (CCAPI), session application (SSAPP), and interactive voice response (IVR).

The following information can be found in the Cisco IOS Debug Command Reference:

- Using debug commands is contained in the "Using Debug Commands" chapter.
- Conditionally triggered debugging is contained in the "Conditionally-Triggered Debugging" chapter.
- Details about individual debug commands are listed.

The following articles contain information about voice debug:

- Voice Debug Concepts
- Voice Call Debug Output Management
- Enhanced Debug Command Sample Output Examples

Filtering Troubleshooting Output

The methods for filtering troubleshooting output are in the following articles:

- Show and More Command Output Filtering.
- Voice Call Debug Filtering on Cisco Voice Gateways
- Voice Call Debug Filtering on H.323 Gatekeepers
- SIP Debug Output Filtering Support
- MGCP Call Centric Debug

Cisco VoIP Internal Error Codes

The Cisco VoIP Internal Error Codes feature generates internal error codes (IECs) for gateway-detected errors that cause the gateway to release or refuse a call. The following articles are about IECs:

- Cisco VoIP Internal Error Codes
- IEC Configuration Examples
- VoIP Network Troubleshooting Using IECs

Troubleshooting Cisco IOS Voice Telephony

If you are troubleshooting a connection to a PBX, you might find the PBX interoperability notes useful. These notes contain configuration information for Cisco gateways and several types of PBXs. To access

these notes, use the following website:
http://www.cisco.com/en/US/netsol/ns728/networking_solutions_program_category_home.html

Analog Voice Interface Troubleshooting

If you are troubleshooting an analog connection, you must understand what type of circuit and interface your voice port is using. 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.

Analog voice telephony interfaces include foreign exchange office (FXO), foreign exchange station (FXS), and receive and transmit (E&M). Direct Inward Dialing (DID) is a service offered by telephone companies that enables callers to dial directly to an extension on a PBX without the assistance of an operator or automated call attendant.

To troubleshoot analog voice interfaces, see the following articles:

- FXS Interfaces
- FXO Interfaces
- E&M Interfaces
- Analog DID Interfaces
- Analog Voice Port Testing Commands

Digital Voice Interface Troubleshooting

Digital voice ports are found at the intersection of a packet voice network and a digital, circuit-switched telephone network. Digital voice telephony interfaces include T1 or E1 channel-associated signaling (CAS), ISDN primary-rate interface (PRI) or basic rate interface (BRI), and E1 R2 signaling.

To troubleshoot digital voice interfaces, see the following articles:

- Digital Voice Interface Hardware Troubleshooting
- Digital Signal Processor Troubleshooting
- Codec Complexity Verification
- T1 or E1 Interface Troubleshooting
- Digital Voice-Port Configuration Verification
- Voice Port Testing Commands

VoIP Quality of Service Troubleshooting

The primary goal of Quality of Service (QoS) is to make network service better and more predictable by providing dedicated bandwidth, controlled jitter and latency, and improved loss characteristics. QoS achieves these goals by providing tools for managing network congestion, shaping network traffic, using expensive wide-area links more efficiently, and setting traffic policies across the network.

For information about configuring QoS for voice, refer to the Quality of Service for Voice document. For more information about QoS, refer to the Cisco IOS Quality of Service Solutions Configuration Guide.

For QoS troubleshooting, see the following articles:

- VoIP QoS Issues
- Common QoS Problems

* Voice Call Tuning

Troubleshooting Cisco IOS Voice Protocols

H.323 Interface Troubleshooting

These articles provide troubleshooting information for the H.323 standard from the International Telecommunication Union Telecommunication Standardization Sector (ITU-T), of the following:

* Cisco H.323-compliant gatekeeper
* Cisco H.323-compliant gateway
* Cisco H.323-compliant features

Cisco IOS software complies with the mandatory requirements and several of the optional features of the H.323 Version 2 specification. The following articles describe H.323 features:

* H.323-Related Standards
* H.323 Gateway Troubleshooting
* H.323 Gatekeeper Troubleshooting
* H.323 Gateway-to-Gateway and Gatekeeper-to-Gateway Security

H.323 is a peer-to-peer protocol. H.323 uses messages similar to Q.931 messages to communicate between endpoints. Q.931 cause codes can be found in SIP and H.323 Internal Cause Codes.

SIP Interface Troubleshooting

The Cisco Session Initiation Protocol (SIP) implementation enables Cisco access platforms to signal the setup of voice and multimedia calls over IP networks. SIP is an ASCII-based, application-layer control protocol that can be used to establish, maintain, and terminate calls between two or more endpoints. SIP is an alternative protocol developed by the Internet Engineering Task Force (IETF) for multimedia conferencing over IP. SIP features are compliant with IETF RFC 2543, SIP: Session Initiation Protocol, published in March 1999. You can view RFC 2543 at http://www.ietf.org/rfc/rfc2543.txt.

Like other Voice-over-IP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

The following articles provide procedural and reference information that you can use to determine and resolve problems with SIP interfaces to the IP network.

Note: Call flows can help in troubleshooting SIP problems. SIP call flow information can be found in the Cisco IOS SIP Configuration Guide.

SIP troubleshooting can be found in the following articles:

* Cisco SIP IP Phone 7960 Troubleshooting
* Cisco SIP Gateway Troubleshooting
* Cisco SIP Proxy Server Troubleshooting
* SIP Messages and Methods

MGCP and Related Protocol Interface Troubleshooting

Media gateway control protocol (MGCP) bases its call control and intelligence in centralized *call agents*, also called media gateway controllers. The call agents issue commands to simple, low-cost endpoints, which are housed in media gateways (MGs), and they also receive event reports from the gateways. MGCP messages between call agents and media gateways are sent over IP/UDP.

For information about configuring MGCP, refer to the MGCP and Related Protocols Configuration Guide.

Use the following articles to troubleshoot MGCP:

- MGCP Overview
- MGCP Call Routing and Dial Peers
- MGCP Call Admission Control
- MGCP Connections and Endpoints Verification
- MGCP Testing Commands

Voice over Frame Relay Interface Troubleshooting

Voice over Frame Relay (VoFR) enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network, using the FRF.11 protocol. This protocol specification defines multiplexed data, voice, fax, dual-tone multifrequency (DTMF) digit-relay, and channel-associated signaling (CAS).

The Cisco VoFR implementation enables you to make dynamic- and tandem-switched calls and Cisco trunk calls. Dynamic-switched calls have dial-plan information included that processes and routes calls based on the telephone numbers. The dial-plan information is contained within dial-peer entries.

VoFR articles include the following topics:

- Frame Relay Troubleshooting Tasks
- VoFR with QoS Troubleshooting
- VoIP over Frame Relay with Multipoint PVCs and Prioritization Troubleshooting
- VoFR Testing Commands

Voice over ATM Interface Troubleshooting

Voice over ATM (VoATM) enables a router to carry voice traffic (for example, telephone calls and faxes) over an ATM network. To troubleshoot VoATM issues, see the following article:

- Voice over ATM Troubleshooting

Troubleshooting Cisco IOS Telephony Applications

Voice Application Troubleshooting

Tcl and VoiceXML applications on the Cisco gateway provide Interactive Voice Response (IVR) features and call control capabilities such as call forwarding and voice mail.

The Cisco voice gateway allows voice applications to be used during call processing. Typically, application scripts contain both executable files and audio files that interact with the system software. Tcl scripts and VoiceXML documents can be stored in any of the following locations: TFTP, FTP, or HTTP servers, Flash memory of the gateway, or on the removable disks of the Cisco 3600 series. The audio files that they reference can be stored in any of these locations, and on Real Time Streaming Protocol (RTSP) servers. A Cisco voice gateway can have several voice applications to accommodate many different services, and you can customize the voice applications to present different interfaces to the various callers. IP phones can also originate calls to a gateway running a voice application.

Voice applications on Cisco gateways can be developed using a choice of two scripting languages:

- Tcl IVR 2.0-Tcl-based scripting with an API.
- VoiceXML-Standards-based markup language for voice browsers.

Applications can also be developed using a hybrid of both Tcl and VoiceXML. The following articles describe troubleshooting Cisco IOS Tcl and VoiceXML applications:

- Tcl IVR Troubleshooting
- Media Inactive Call Detection
- Cisco VoiceXML Troubleshooting
- Tcl IVR Events and Status Codes

AAA and Billing Application Troubleshooting

To troubleshoot authentication, authorization, and accounting (AAA), billing, and settlement issues for voice services, refer to the following articles:

- AAA for Voice Troubleshooting
- Accounting Server Connectivity Failure and Recovery Detection
- Enhanced Billing Support for SIP Gateways Troubleshooting
- Settlement Troubleshooting

Fax Application Troubleshooting

This chapter describes T.37 store and forward fax and T.38 fax gateway troubleshooting concepts. The applications are T.37 store and forward fax, T.38 fax relay for VoIP H.323, fax relay packet loss concealment, and T.37/T.38 fax gateways. The applications enable the Cisco gateways to send and receive faxes across packet-based networks, using modems or voice feature cards (VFCs).

Before beginning these troubleshooting procedures, check to ensure your fax machine and voice network are working. Try the following steps to help isolate the problem:

1. Plug the fax machine into a regular analog line and test the fax call while bypassing the voice network. If the problem persists, it could be the fax machine.
2. If the fax machine has a handset, connect the fax machine to the voice network, pick the handset up, and try to make a voice call. If the voice call works, you have verified that your voice network is operational. A voice call must be successful before a fax call can succeed.

Fax troubleshooting includes the following articles:

- Fax Call Flow
- Fax Relay
- Fax Detection

For information about configuring fax features, refer to the Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide.

Monitoring the Cisco IOS Voice Network

To monitor your voice network, use the following articles:

- Voice Network Monitoring
- Voice Performance Statistics on Cisco Gateways
- Configuring Voice Performance Statistics on Cisco Gateways
- Managing the Collection of Voice Statistics
- Voice Performance Statistics on Cisco Gateways Configuration Examples

Cause Codes and Debug Values

The following articles contain tables of cause codes and debug values:

- VoIP Cause Codes and Debug Values
- SIP and H.323 Internal Cause Codes
- Tcl IVR Events and Status Codes