This article describes troubleshooting features and tips for the Cisco SIP IP phone 7960.

| Guide Contents |
|---|
| Troubleshooting Cisco IOS Voice Overview |
| Debug Command Output on Cisco IOS Voice Gateways |
| Filtering Troubleshooting Output |
| Cisco VoIP Internal Error Codes |
| Troubleshooting Cisco IOS Voice Telephony |
| Troubleshooting Cisco IOS Voice Protocols |
| Troubleshooting Cisco IOS Telephony Applications |
| Monitoring the Cisco IOS Voice Network |
| Cause Codes and Debug Values |

# Contents



# Troubleshooting Features

The following is a list of features on the Cisco SIP IP phone that you can use for troubleshooting:

- Settings button to Network Configuration soft key-Use to view or modify the network configuration of the phone.
- Settings button to SIP Configuration soft key-Use to view or modify a phone's SIP settings.
- Settings button to Status-Display configuration or initialization errors.
- Call messages on LED screen-Display basic SIP message flows.
- Pressing *i* or **?** key twice during a call-Displays real-time transferring and receiving call statistics. This option is recommended for troubleshooting voice-quality issues.

In addition to the features listed above, the EIA/TIA-232 (RS-232) port located on the back of the Cisco SIP IP phone 7960 is a console port and can be used to gather debug information.

The EIA/TIA-232 port is password-protected and requires a custom RJ-11-to-RJ-45 cable.

**Note**: For a PC connection, the RJ-45 connection needs a DB-9 female DTE adapter or an RJ-45 crossover cable for an octal async connection. You must enter the password "cisco" must be entered to enable any output to be seen via the EIA/TIA-232 port. The connection baud rate, parity, start bits, and stop bits are 9600, N, 8, and 1.

To use the console port, use a RJ-11-to-RJ-45 custom cable to connect the EIA/TIA-232 port to a PC.

Table: RJ-11-to-RJ-45 Pinouts lists the RJ-11-to-RJ-45 cable pinouts.

**Table: RJ-11-to-RJ-45 Pinouts**

| RJ-11 or RJ-12 | RJ-45 |
|---|---|
| 2 | 6 |
| 3 | 4 |
| 4 | 3 |

To connect the console port, complete the following tasks:

1. Insert the RJ-11 end of the rolled cable into the EIA/TIA-232 port on the back of the phone.
2. Use an RJ-45-to-DB-9 female DTE adapter (labeled TERMINAL) to connect the console port to a PC running terminal emulation software.
3. Insert the RJ-45 end of the rollover cable into the DTE adapter.
4. From the console terminal, start the terminal emulation program.
5. Type "cisco". A prompt is displayed.
6. At the prompt, you can issue the following commands to assist you in troubleshooting and debugging the phone:

- **debug error**-Displays error messages that are occurring in the call flow process
- **debug sip-message**-Enables you to view a text display of a call flow

# Troubleshooting Tips

This section provides tips for resolving the following Cisco SIP IP phone problems:

- Cisco SIP IP Phone Is Unprovisioned or Is Unable to Obtain an IP Address
- Cisco SIP IP Phone Does Not Register With the SIP Proxy or SIP Registrar Server
- Outbound Calls Cannot Be Placed from a Cisco SIP IP Phone
- Inbound Calls Cannot Be Received on a Cisco SIP IP Phone
- Poor Voice Quality on the Cisco SIP IP Phone
- DTMF Digits Do Not Function Properly
- Cisco SIP IP Phones Do Not Work When Plugged into a Line-Powered Switch
- Call Transfer Does Not Work Correctly
- Some SIP Messages are Retransmitted Too Often

For more information about Cisco SIP IP phones, see the Cisco IP Phone Administrator Guides for SIP.

## Cisco SIP IP Phone Is Unprovisioned or Is Unable to Obtain an IP Address

To determine why a phone is unprovisioned or unable to obtain an IP address, perform the following tasks as necessary:

- If using TFTP to download configuration files, verify that the SIPDefault.cnf file and the phone-specific configuration file (SIP*mac*.cnf where *mac* is the MAC address of the phone) exist and are configured correctly.
- Verify that the TFTP server is working properly.
- Verify that the Cisco SIP IP phone network configuration parameters are properly configured and the phone is obtaining the proper IP addressing information (IP address, subnet mask, default gateway, TFTP server, and so forth.)

- Press the **Settings** button, select **Status**, and then **Status Messages** to view messages for missing files or other errors.
- If the DHCP server has an IP subnet mask that is different from the one for the Cisco SIP IP phone, verify that "ip helper-address" address is enabled on the local router.
- Verify that the Cisco SIP IP phone software image (P0S3*xxyy*.bin where *xx* is the version number and *yy* is the subversion number) was downloaded from the Cisco website in binary format.

## Cisco SIP IP Phone Does Not Register With the SIP Proxy or SIP Registrar Server

To determine why a phone does not register with a SIP proxy or SIP registrar server, perform the following tasks as necessary:

**Note**: The character "x" displayed to the right of a line icon indicates that registration has failed.

- Verify that phone registration with a proxy server is enabled (via the proxy_register parameter in the configuration files). By default, registration during initialization is disabled.
- Verify that the IP address (proxy1_address parameter) of the primary SIP proxy server to be used by the phones is valid.
- If a Fully Qualified Domain Name (FQDN) is specified in the proxy1_address parameter, verify that the DNS server is configured to resolve the FQDN as a DNS A-record type.
- Verify that the Cisco SIP proxy server has been configured to require authentication. If it has, ensure that an authentication name and password have been defined in the Cisco SIP IP phone-specific configuration file (through the use of the linex_authname and linex_password parameters).
- The Cisco SIP IP phone currently supports the HTTP Digest authentication method. Verify that the authentication method required by the Cisco SIP proxy server (through the use of the AuthScheme directive in the sipd.conf file) is HTTP Digest.
- Verify that a registration request hasn't expired. By default, Cisco SIP IP phones reregister every 3600 seconds, but this value can be modified through the use of the time_reqister_expires parameter.

## Outbound Calls Cannot Be Placed from a Cisco SIP IP Phone

If a call cannot be placed from a Cisco SIP IP phone, perform the following tasks as necessary:

- Verify that the Cisco SIP IP phone network configuration IP address parameters have been correctly entered or received from a DHCP server.
- Verify that the Cisco SIP proxy server used by the phone is working properly.
- Verify that the Cisco SIP proxy server is correctly configured for routes or registrations to the remote destination.
- Verify that the remote SIP device is available.
- Verify that a dial plan has been defined in the dialplan.xml file and if so, that the configuration is correct. This file should have been downloaded from CCO to the root directory of your TFTP server.
- Determine which error tones are being received and map those tones to the messages displayed on the phone's LCD (SIP 4xx messages, and so forth.)

## Inbound Calls Cannot Be Received on a Cisco SIP IP Phone

If inbound calls cannot be received on a Cisco SIP IP phone, perform the following tasks as necessary:

- Verify that the line (user portion) was defined in the Request-URI or the SIP INVITE request. The Cisco SIP IP phone requires this information to determine the proper line to ring.
- Verify that the Request-URI is sent to port 5060 of the phone's IP address. The phone listens on UDP port 5060.

- Verify that the Cisco SIP IP phone is registered with the local proxy server.

## Poor Voice Quality on the Cisco SIP IP Phone

If a call's voice quality is compromised on the Cisco SIP IP phone, perform the following tasks as necessary:

- Check the network path for errors, packet drops, loss, loops, and so forth.
- Verify that the ToS level for the media stream being used has been correctly set (through the tos_media parameter in the configuration file).
- Verify that the Cisco SIP IP phone is plugged into a switch rather than a hub to avoid excessive collisions and packet loss.
- Ensure that there is enough bandwidth on the network for the selected codec (especially for calls over a WAN).
- Press the **i** or **?** button twice on the phone during the call to view realtime transferring and receiving call statistics.
- Determine whether the problem occurs with the handset, headset, or speaker phone, or with all of them.

## DTMF Digits Do Not Function Properly

If DTMF digits are not functioning properly, perform the following tasks as necessary:

- If out-of-bound signaling through the AVT tone method has been enabled (through the dtmf_outofband configuration file parameter), verify that the remote device supports AVT tones (as defined in RFC 2833). If AVT tones have been enabled and the remote device does not support AVT tones, check for packet loss in the end-to-end path.
- Find out which codec is being used. Lower bandwidth codecs yield poorer results if AVT tones are not supported because the DTMF digits are carried in audio.
- Verify the length of the tones being created. The tone must have a minimum signal duration of 40 ms with signaling velocity (tone and pause) of no less than 93 ms (as defined in RFC 2833).

## Cisco SIP IP Phones Do Not Work When Plugged into a Line-Powered Switch

If the Cisco SIP IP phones do not work when plugged into a line-powered switch, perform the following tasks:

- Verify that the phone is running version 2.0 or higher of the Cisco SIP IP Phone software. (Line-powered support was not available in version 1.0.)
- Verify that the network media type Network Settings parameter is set to auto-negotiation (auto).

## Call Transfer Does Not Work Correctly

If call transfer does not work correctly, verify that the remote SIP device that is sending the call is using the SIP BYE/Also: method (as defined in Internet draft sip-cc-01.txt.)

## Some SIP Messages are Retransmitted Too Often

The Cisco SIP IP phone has several timers (INVITE request retries, BYE request retries, etc.) that can be configured using the sip_invite_retx and sip_retx configuration file parameters. In most networks, the default values work fine, however, conditions such as network delay, slower-processing proxy servers, and packet loss might require that the timers be adjusted. If some SIP messages appear to be retransmitted too often, adjust these parameters.