

**SoftCo VoIP Integrated Exchange  
V100R002C04SPC600  
Troubleshooting**

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# 1 Overview

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## About This Chapter

This topic describes the fault categories, precautions of troubleshooting, fault information sources, overall troubleshooting process and how to obtain help of huawei when you can not deal with the fault.

### 1.1 Fault Level

This topic describes how to rectify faults at different levels.

### 1.2 Fault Categories

The troubleshooting method varies according to fault category.

### 1.3 Precautions

Before locating and troubleshooting faults, maintenance personnel must read and comply with the relative precautions.

### 1.4 Fault Sources

SoftCo faults are found in customer complaints, routine maintenance, and alarms.

### 1.5 Troubleshooting Process

Before proceeding, learn the basic troubleshooting process.

### 1.6 Technical Support

Huawei Technologies Co., Ltd. provides customers with comprehensive technical support and service. Please feel free to contact our local office or company headquarters.

## 1.1 Fault Level

This topic describes how to rectify faults at different levels.

Depending on the impact and scope of the fault, faults are classified as either emergency faults or ordinary faults.

Emergency faults refer to those that occur suddenly and affect a wide range of services or devices. Emergency faults, such as host breakdown and global service congestion, seriously affect network operations and the quality of service (QoS).

The following faults are emergency faults:

- All or the most majority of SoftCo functions are unavailable.
  - A single board or all boards are unavailable.
  - The SoftCo is powered off.
- All SoftCo services are globally interrupted for a long time or calls are congested.
  - None of the media gateways can register with the network, for example, a media gateway is disconnected immediately after it is registered.
  - Calls are congested at all user terminals. For example, users cannot make or receive calls, or cannot hear the dial tone or busy tone after picking up the phone.
  - Calls are congested at all office directions. For example, users fail to make or receive calls.
- Some SoftCo services are interrupted for a long time or calls are congested.
  - Some media gateways fail to register with the network, or disconnect immediately after they are registered.
  - Call congestion occurs in some user terminals. For example, users cannot make or receive calls, or cannot hear the dial tone or busy tone after picking up the phone.
  - Call congestion occurs in some office directions. For example, users fail to make or receive calls.

Ordinary faults refer to those faults that are not emergency faults. The policies for locating and rectifying faults and the applicable reference documents are different for emergency and ordinary faults.

**Table 1-1** Policies for locating and rectifying faults

Fault Level	Processing Policy	Reference
Ordinary fault	Locate and rectify the fault immediately.	See <b>SoftCo Product Document &gt; Fault Management &gt; Troubleshooting</b> .
Emergency fault	Restore the service that has been affected as soon as possible, and then find the root cause of the fault.	See <b>SoftCo Product Document &gt; Fault Management &gt; Emergency Maintenance</b> .

## 1.2 Fault Categories

The troubleshooting method varies according to fault category.

Currently, unified communications solution faults of huawei are classified into the following categories:

- [3 Basic Call Faults](#)
- [4 Trunk Faults](#)
- [5 Service faults](#)
- [6 System Device Faults](#)

- [7 Terminal and gateway faults](#)

## 1.3 Precautions

Before locating and troubleshooting faults, maintenance personnel must read and comply with the relative precautions.

- Receive training before troubleshooting faults to gain knowledge about operational regulations and industry security regulations.
- Understand basic operational skills and emergency handling techniques for the equipment.
- Take ESD measures, for example, wear ESD wrist straps, when replacing and maintaining device parts.
- Record original information in detail when any problem occurs during troubleshooting, such as troubleshooting step, symptom description, alarm information and log information.
- Record all major operations, such as system restart and software loading. Confirm the feasibility of an operation before performing it. Do not attempt any operation before taking proper backup, emergency, and security measures.
- Modify the data under authorization and control. Back up the data before modifying it, and record the modified content.
- Never run third-party software to query or modify the database.

## 1.4 Fault Sources

SoftCo faults are found in customer complaints, routine maintenance, and alarms.

### Customer Complaints

When receiving a complaint from a customer, ask the customer to:

1. Describe the fault symptom.
2. Check the parameter settings.
3. Use the UCMaint to capture log information and send it to Huawei technical support.

### Routine Maintenance

Routine maintenance is a set of preventive measures taken regularly during the normal running of the system. The maintenance personnel detect and eliminate potential defects or troubles at the earliest time to ensure sustained safety and system stability.

During routine maintenance, check for the following:

- Whether an alarm is reported.
- Whether the call detail record (CDR) pool is full.
- Whether the trunk link is normal.
- Whether the board is running normally.
- Whether the service network port is working normally.

- Whether all users are in normal state.



**NOTE**

For details, see Routine Maintenance.

## Alarm Information

Device alarms reflect a device's running status. When a major or critical alarm is reported, the device is faulty or potentially faulty and must be cleared immediately.

The alarm information includes the following:

- The board is faulty.
- The CDR pool is full.
- The trunk link is faulty.
- The network is faulty.
- The E1 trunk is faulty.



**NOTE**

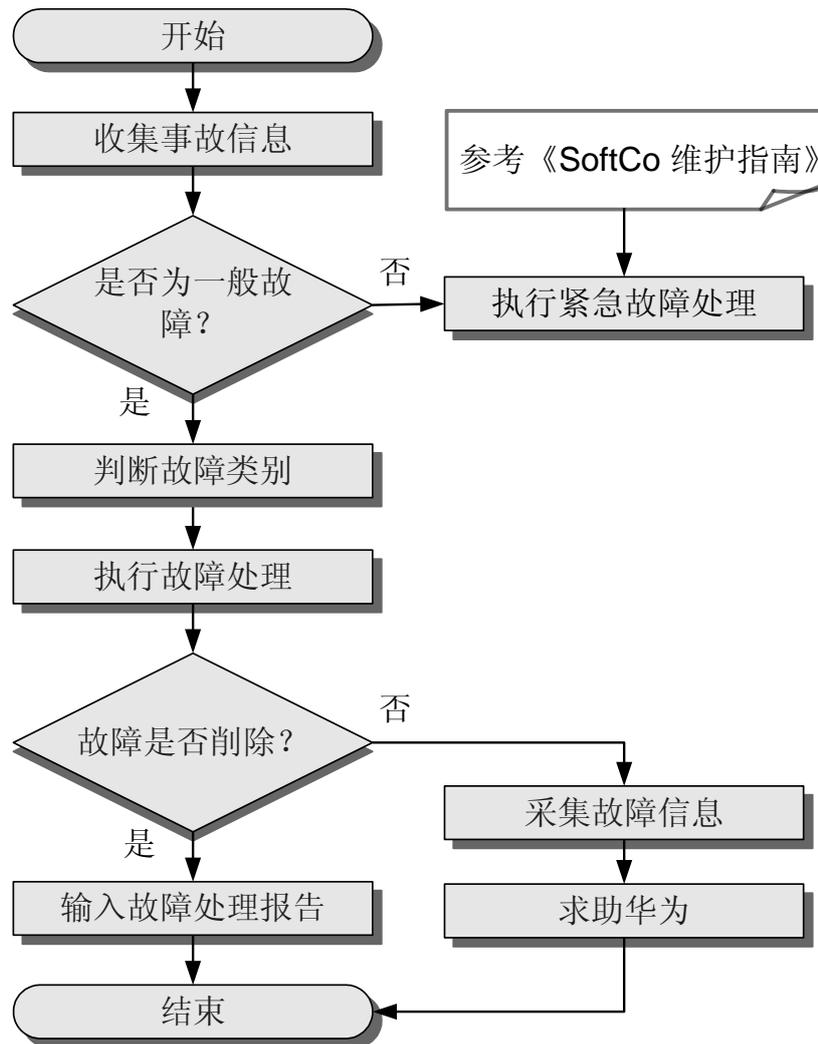
To view the cause of a fault and methods for rectifying the fault, use the Operation and Maintenance Unit (OMU).

## 1.5 Troubleshooting Process

Before proceeding, learn the basic troubleshooting process.

[Figure 1-1](#) shows the overall troubleshooting process.

Figure 1-1 Troubleshooting process



## Collecting Fault Scenario Information

Clear fault scenario information helps to quickly locate faults. When a fault occurs, collect the following information:

- Time and location
- Description of the fault symptom.
- Operations that a user or a maintenance engineer performed before the fault occurred.
- Measures that have taken after the fault occurred and the results.
- Services that were affected by the fault, including the fault impact and scope.

## Checking Whether the Fault is an Normal fault

Upon receiving the fault information, determine whether the fault is an normal fault.

- If yes, locate and rectify the fault according to this document.

- If no, restore the services according to the **SoftCo Product Document > Fault Management > Emergency Maintenance**.

## Judging Fault Categories

Judge the fault categories according to the fault symptoms.

## Performing Troubleshooting

Performing the troubleshooting is composed of locating the fault and rectifying the fault. Locating a fault uses to find the fault reason, and the rectifying the fault uses to remove the fault which restore the device to run normally.

## Verifying That a Fault Is Removed

After measures are taken, verify that the fault symptoms have disappeared.

## Contacting Huawei Technical Support for Help

If the fault persists after you locate the fault and rectify it according to this document, contact Huawei technical support for remote or onsite help. For details on how to obtain Huawei technical support, see Huawei Technical Support.

Before seeking help, ensure that the following information is available:

- Full name of the office where the fault occurred
- Contact information, including a contact person name and a fixed-line phone number or a mobile number
- Fault scenario and symptoms
- Remote maintenance environment and remote access parameters

## Writing a Troubleshooting Report

After verifying that the fault has been rectified, record the troubleshooting process and write a troubleshooting report.



### NOTE

A recommended troubleshooting report consists of four topics: fault symptom, fault locating, fault rectification, and preventive suggestions.

## 1.6 Technical Support

Huawei Technologies Co., Ltd. provides customers with comprehensive technical support and service. Please feel free to contact our local office or company headquarters.

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Email: support@huawei.com

# 2 Common Methods for Locating Faults

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## About This Chapter

The unified communications (UC) solution provides various methods to locate faults.

### 2.1 Data Configuration Analysis

Use the UCMaint to collect data configuration information which locate the configuration fault.

### 2.2 Alarm Analysis

An alarm is an important sign for a fault or an event. Alarms help detect faults in a timely manner, and provide suggestions for troubleshooting.

### 2.3 Log Analysis

Logs record key events and information that are generated while programs are running. By viewing and analyzing logs, you can locate and rectify faults.

### 2.4 Signaling Analysis

Signaling records the process of establishing a call. By analyzing the signaling, you can find out possible causes of call failure and rectify the fault.

### 2.5 Network Information Analysis

This topic describes how to obtain network information.

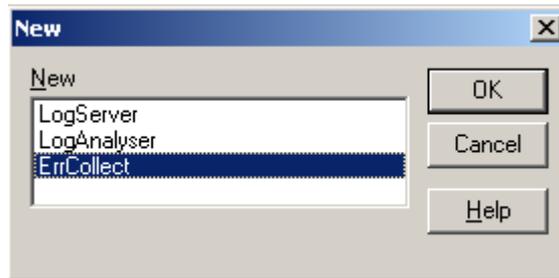
## 2.1 Data Configuration Analysis

Use the UCMaint to collect data configuration information which locate the configuration fault.

Use the UCMaint to collect information through one click. The operation is as follow:

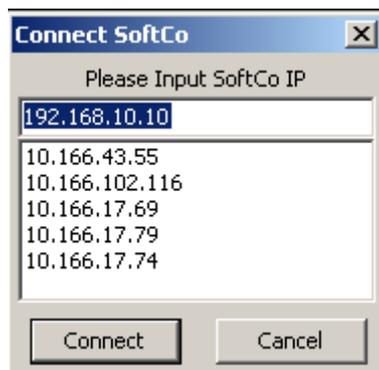
1. Create the **ErrCollect**file.
  - a. In the main menu, choose **File > New**, the **New** dialog is displayed, as shown in [Figure 2-1](#).

**Figure 2-1** New



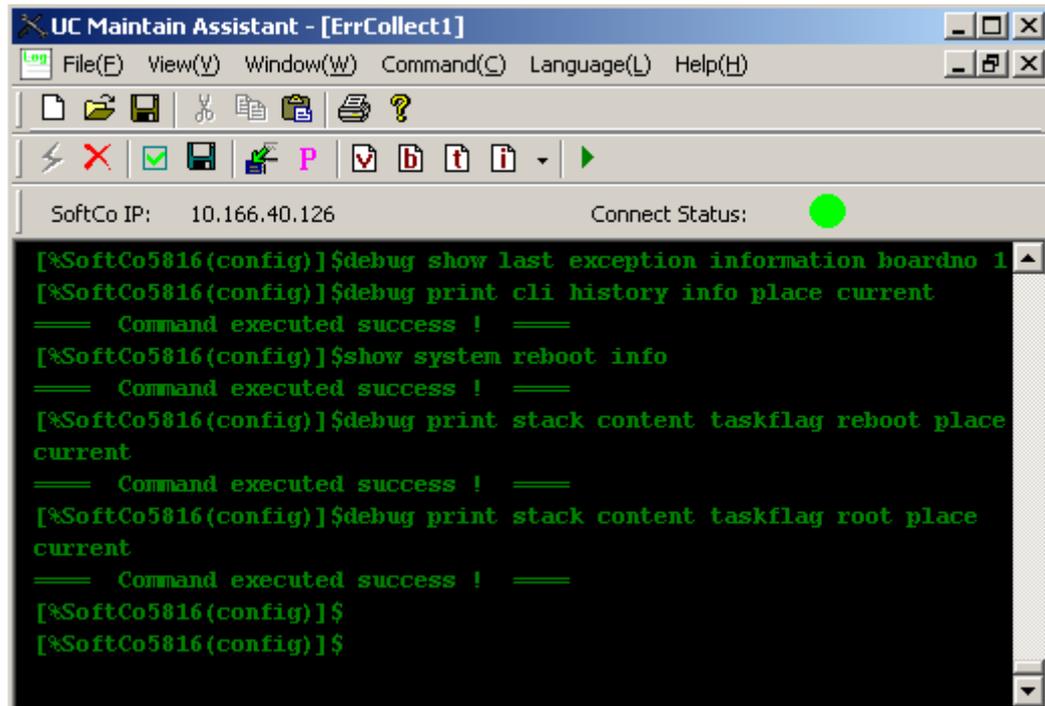
- b. In the **New** list-dialog, choose **ErrCollect**, then click **OK**.
  2. Connect SoftCo.
    - a. In the toolbar, click , the **Connect SoftCo** dialog is displayed, as shown in [Figure 2-2](#).

**Figure 2-2** Connect SoftCo



- b. Input SoftCo IP address, click **Connect**.
  3. In the main window, click , the related configure data will displayed down the main window, as shown in [Figure 2-3](#).

You view whether the data is right, such as the board status, trunk link, and so on.

**Figure 2-3** Data configuration information

The screenshot shows a window titled "UC Maintain Assistant - [ErrCollect1]". The window has a menu bar with "File(F)", "View(V)", "Window(W)", "Command(C)", "Language(L)", and "Help(H)". Below the menu bar is a toolbar with various icons. The main area of the window displays a terminal window with the following text:

```
SoftCo IP: 10.166.40.126          Connect Status: ●  
[%SoftCo5816(config)]$debug show last exception information boardno 1  
[%SoftCo5816(config)]$debug print cli history info place current  
==== Command executed success ! ====  
[%SoftCo5816(config)]$show system reboot info  
==== Command executed success ! ====  
[%SoftCo5816(config)]$debug print stack content taskflag reboot place  
current  
==== Command executed success ! ====  
[%SoftCo5816(config)]$debug print stack content taskflag root place  
current  
==== Command executed success ! ====  
[%SoftCo5816(config)]$  
[%SoftCo5816(config)]$
```

## 2.2 Alarm Analysis

An alarm is an important sign for a fault or an event. Alarms help detect faults in a timely manner, and provide suggestions for troubleshooting.

The priority for rectifying faults depends on the alarm severity.

- **Critical:** global alarms that seriously affect device operations, for example, a power supply fault. Critical alarms must be handled immediately; otherwise, the system may crash.
- **Major:** board or physical circuit alarms that occur in a certain scope. For example, the fault of the physical circuit. Major alarms must be handled immediately; otherwise, the services cannot run normally.
- **Minor:** general faults or event alarms that describe the working status of each board or physical circuit. You must locate the cause to remove potential problems.
- **Warning:** prompt message, for example, a message indicating that a device has been restored. Warning alarms do not affect system performance and services. No action is required.



### NOTE

If the Operation and Maintenance Unit (OMU) or Unified Communications Element Management System (UCEMS) is used, the alarm severity is displayed in the alarm window.

The procedure for handling an alarm is as follows:

1. Find the critical and major alarms among all alarms.
2. View the cause of each alarm and handle the alarm.

3. Verify that the alarm is cleared.
  - If the alarm is cleared, the fault is rectified.
  - If the fault is not rectified, find out other possible causes.

To view alarms, use the OMU or UCEMS, or observe the indicators.

- Observe the SoftCo indicators to obtain alarm information.

Table 2-1 and Table 2-2 describe the indicators.

**Table 2-1** Indicator status on the SoftCo chassis

Indicator	Color	Status Description
POWER	Green	<ul style="list-style-type: none"> <li>• On: The power supply is normal.</li> <li>• Off: There is no power.</li> </ul>
RUN	Green	<ul style="list-style-type: none"> <li>• Blinking (at 4 Hz): The board is writing data to the FLASH memory while the system is running.</li> <li>• Blinking (at 0.5 Hz): The system is running normally.</li> <li>• On: The board is faulty.</li> <li>• Off: There is no power, the board is faulty, or the system is starting.</li> </ul>
ALARM	Red	Applicable only to the SoftCo9500. <ul style="list-style-type: none"> <li>• Blinking: An alarm is being reported.</li> <li>• Off: No alarm is being reported.</li> </ul>

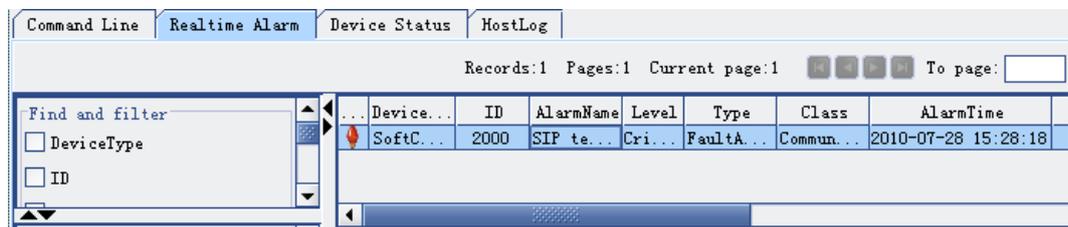
**Table 2-2** Board status indicators

Indicator	Color	Status Description
RUN	Green	<ul style="list-style-type: none"> <li>• Blinking (at 4 Hz): The board is writing data to the FLASH memory while the system is running.</li> <li>• Blinking (at 2 Hz): The board is writing data to the FLASH memory during system startup.</li> <li>• Blinking (at 1 Hz): The board is starting. It is applicable only to Main Control Units (MCUs).</li> <li>• Blinking (at 0.5 Hz): The system is running normally.</li> <li>• On: The board is faulty.</li> <li>• Off: There is no power, or the board is faulty.</li> </ul>
ALARM	Red	For MCUs: <ul style="list-style-type: none"> <li>• Blinking: An alarm is being reported while the system is running.</li> <li>• Off: No alarm is being reported.</li> </ul> For other boards:

Indicator	Color	Status Description
		<ul style="list-style-type: none"> <li>• <b>Blinking:</b> The board is starting, or the user does not add the board by using a command after the SoftCo has started.</li> <li>• <b>Off:</b> The board is running normally.</li> </ul>
ACT (active/standby indicator)	Green	Applicable only to the MCUs. <ul style="list-style-type: none"> <li>• <b>On:</b> This board is the active board.</li> <li>• <b>Off:</b> This board is the standby board.</li> </ul>
LOS (signal loss indicator)	Yellow	Applicable only to Digital Trunk Units (DTUs). <ul style="list-style-type: none"> <li>• <b>On:</b> Signals are lost.</li> <li>• <b>Off:</b> No signal is lost.</li> </ul>
RFA (remote alarm indicator)	Green	Applicable only to the DTUs. <ul style="list-style-type: none"> <li>• <b>On:</b> The peer device is not receiving signals.</li> <li>• <b>Off:</b> The peer device is running normally.</li> </ul>

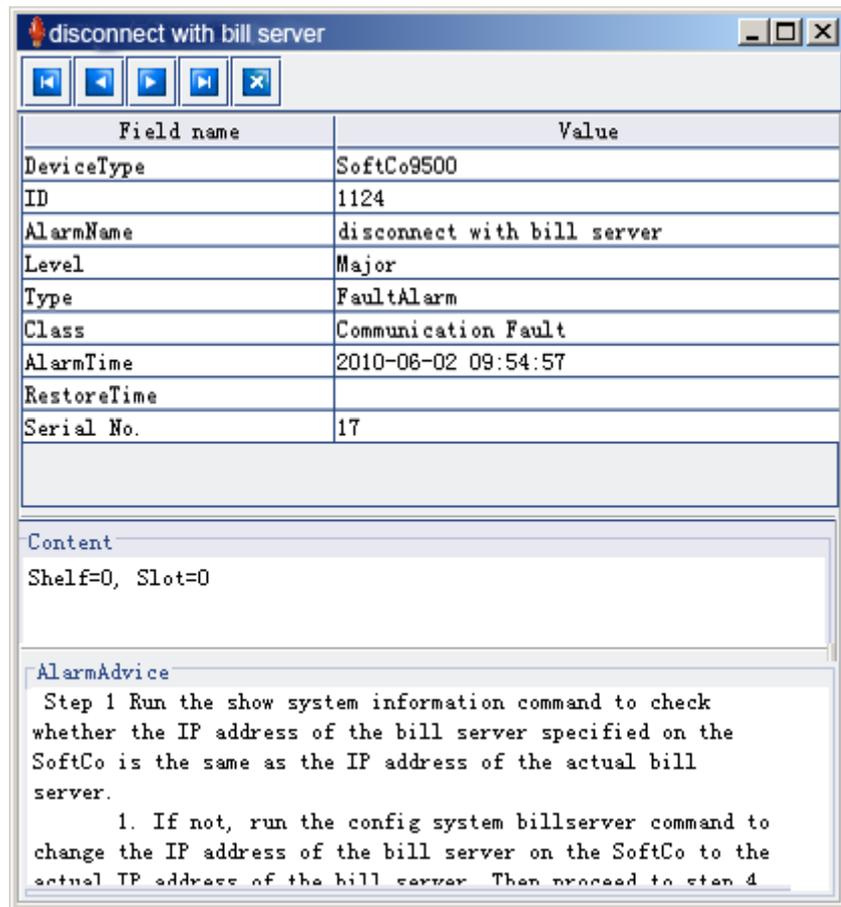
- Use the OMU to view alarm information.
1. On the OMU main page, click the **Realtime Alarm** tab to view alarm information, as shown in [Figure 2-4](#).

**Figure 2-4** Realtime Alarm tab page



2. In the alarm browse area, double-click a required alarm. The detailed alarm information is displayed, as shown in [Figure 2-5](#).

**Figure 2-5** Detailed alarm information



3. Rectify alarms based on the suggestions provided.

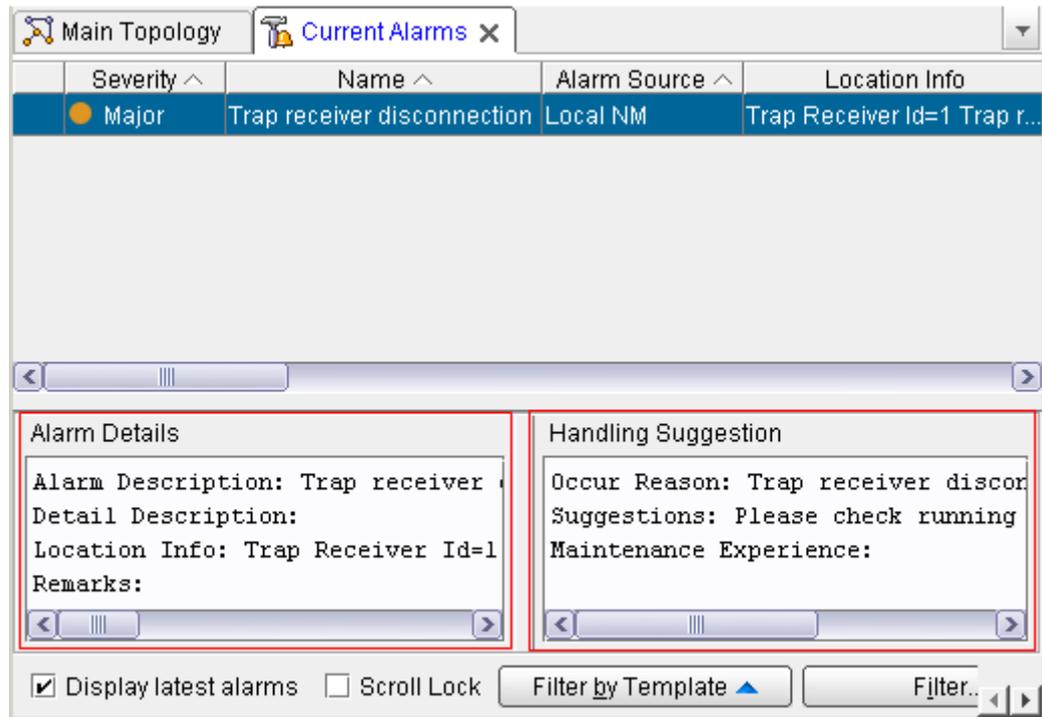
**NOTE**

The OMU installation package is obtained from <http://support.huawei.com>. For details on how to use the OMU, see the *OMU Client Help*.

- Use the UCEMS to view alarm information.
- 1. On the main menu, choose **Fault > Browse Current Alarm**.
- 2. In the alarm browse window, select a required alarm.

The alarm details along with the suggestions are displayed, as shown in [Figure 2-6](#).

**Figure 2-6** Current alarm information



**NOTE**

The UCEMS installation package is obtained from <http://support.huawei.com>. For details on how to use the UCEMS, see the *iManager UCEMS Help*.

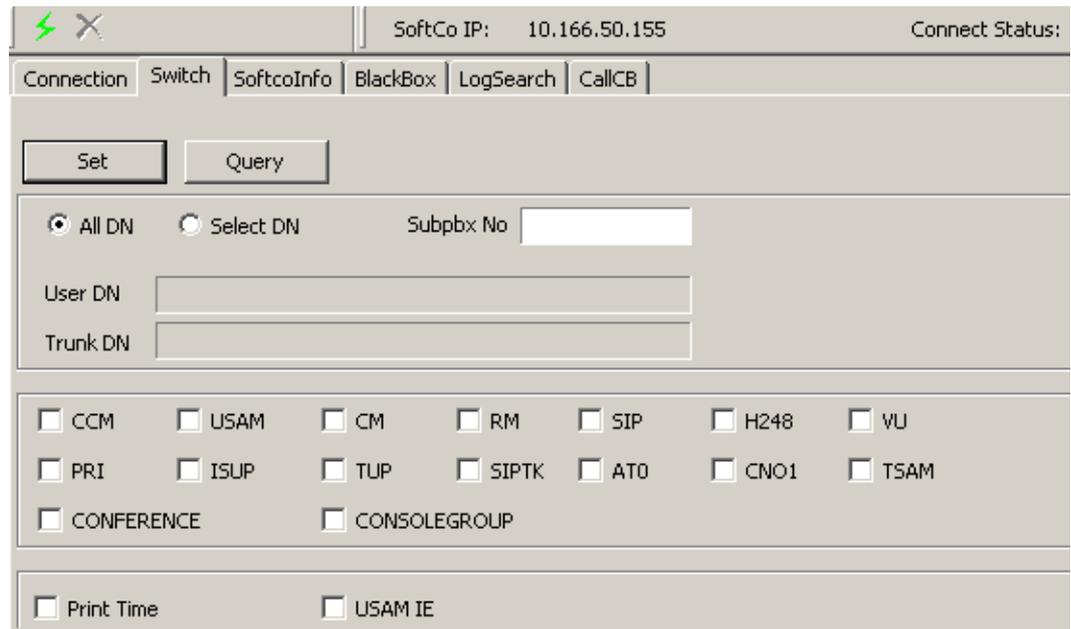
## 2.3 Log Analysis

Logs record key events and information that are generated while programs are running. By viewing and analyzing logs, you can locate and rectify faults.

The unified communications (UC) solution provides the UCMaint to view logs.

1. Click the **Switch** tab. Select required log modules, and click **Set** to enable the log module functions, as shown in [Figure 2-7](#).

**Figure 2-7** Switch tab page



2. Click the **LogSearch** tab. Set parameters such as the calling number and called number, and click **QueryLog**, as shown in [Figure 2-8](#).

**Figure 2-8** LogSearch tab page



3. Double-click a record to view the log details, as shown in [Figure 2-9](#). The calling number 5001 and called number 5002 are contained in the log.

Figure 2-9 Log information

```
20100729143517.log - Text Document.txt
File Edit Format View Help
[sipusm.c 1781][dwUsn= 169] SIPUSH_SetRemoteCRD, from tunnel IE
[sipsrvglobal.c 3251]SIPUtil_SetCRDFromSipMeidaIE, 2 channel num
[sipusm.c 1823][dwUsn= 169] media offer sent
[sipusm.c 1127][dwUsn= 169] SIPUSH_SetUserState FROM [IDLE] TO [
[sipusm.c 7339][dwUsn= 169] caller:5001, callee:5002
[sipusm.c 3791][dwUsn= 169] SIPUSH_AddSipMediaIEForSIPAPP, Get n
[sipusm.c 7551][dwUsn= 169] SIPUSH_SendSetupToSIPAPP succeed, dw
[sipusm.c 1181][dwUsn= 169] [dwCRNo= 4]SIPUSH_SetCRState: fro
[sipsdl.c 9140]SipAppSdlSetMediaEncoding() No Ptime
[sipsdl.c 9140]SipAppSdlSetMediaEncoding() No Ptime
[ccm.c 2486][dwCCMNo= 2] CCM_ProcStateCallMsg
[ccm.c 16691] CCM_ProcStateCallMsg::<- MsgType = EN_Q931_MSG_SE
```

 NOTE

The UCMaint installation package is obtained from <http://support.huawei.com>. For details on how to use the UCMaint, see the *UCMaint User Guide*.

## 2.4 Signaling Analysis

Signaling records the process of establishing a call. By analyzing the signaling, you can find out possible causes of call failure and rectify the fault.

- View signaling information using a packet capture tool.  
For details, see *How to Use the Wireshark*.
- View signaling information using the Operation and Maintenance Unit (OMU).

[Figure 2-10](#) shows the SS7 signaling process.

**Figure 2-10** Signaling tracing

The screenshot shows a signaling tracing interface with the following components:

- Search and Filter:**
  - Begin Time: 2010-08-19 15:17:43
  - End Time: 2010-08-19 15:47:43
  - Direction: ALL
  - CIC: (empty)
  - Message Type: (empty)
  - Caller: (empty)
  - Callee: (empty)
- Table:**

Serial No.	Time	Direction	Model
9	2010-08-12 14:47:05	SEND	ISUP
8	2010-08-12 14:47:05	RECV	ISUP
7	2010-08-12 14:47:03	RECV	ISUP
6	2010-08-12 14:47:01	SEND	MTP
5	2010-08-12 14:47:01	RECV	MTP
4	2010-08-12 14:47:01	RECV	ISUP
3	2010-08-12 14:47:01	SEND	ISUP
2	2010-08-12 14:47:01	RECV	MTP
1	2010-08-12 14:47:01	SEND	MTP
- NO. 7 Signal:**
  - 00001100 + LI
  - 10000101 + SIO
  - + STF

**NOTE**

The OMU installation package is obtained from <http://support.huawei.com>. For details, see the *OMU Client Help*.

## 2.5 Network Information Analysis

This topic describes how to obtain network information.

Running the **ping** command to view network information

- Run the **ping -count<1-65535> waittime<1-255> ttl<1-255> <a.b.c.d>** command on a SoftCo to check the network status. The following is an example which the network status is normal:

```
[%SoftCo9500(config)]#ping -count 10 192.168.10.11

==== Command executed success ! ====

Pinging 192.168.10.11 with 32 bytes of data:

Reply from 192.168.10.11: bytes=32 time<10ms ttl=64
Reply from 192.168.10.11: bytes=32 time<10ms ttl=64
Reply from 192.168.10.11: bytes=32 time<10ms ttl=64
```

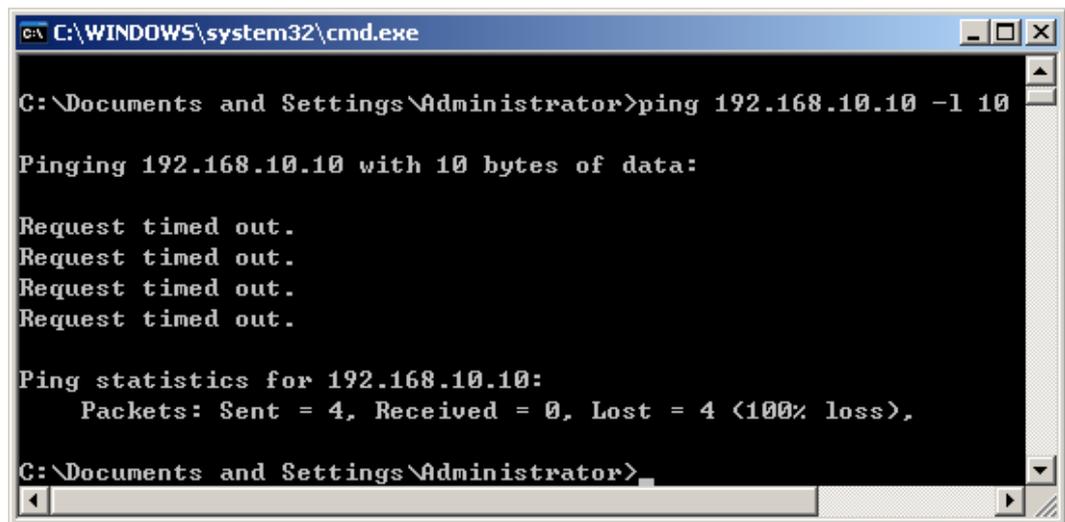
```

Reply from 192.168.10.11: bytes=32  time<10ms  ttl=64
Ping statistics for 192.168.10.11
    Packets: Sent = 10, Received = 10, Lost = 0<0% loss>,
Approximate round trip times in milli-seconds:
    Minimum = 0ms, Maximum = 0ms, Average = 0ms

==== Command executed success ! ====
    
```

- Run the **ping<a.b.c.d> -l<count>** command in the command line interface (CLI) of a personal computer to check the network status.
  1. Choose **Start > Run**, and then enter the **cmd** command in the **Run** dialog box.
  2. Click **OK** to display the CLI, as shown in [Figure 2-11](#).

**Figure 2-11** Cmd



3. Enter **ping** command to check the network status. The abnormal network status is shown in [Figure 2-11](#)

[Table 2-3](#) lists the IP bearer network quality.

**Table 2-3** IP bearer network quality

Network Quality	One-Way Delay (ms)	Packet Loss Rate	Jitter (ms)
Good (recommended)	≤ 40	≤ 0.1%	≤ 10
Poor	≤ 100	≤ 1%	≤ 20
Terrible	≤ 400	≤ 5%	≤ 60

Table 2-4 lists the data service usage that is available based on the IP bearer network quality.

**Table 2-4** Data service usage

Service Type		Good	Poor	Terrible
Fax	Transparent transmission (G.711a/u)	Available	Unavailable	Unavailable
	T.38	Available	Available	Unavailable
Modem	Transparent transmission (G.711a/u)	Available	Unavailable	Unavailable

# 3 Basic Call Faults

---

## About This Chapter

Basic call faults include one-way audio, no voice, low call quality, call connection fault, digit sending and collection fault, number display fault, and prompt tone fault.

### 3.1 Background

This topic describes call principles.

### 3.2 Only One Party Can Hear the Peer Voice

This topic describes how to troubleshoot a fault that occurs when only one party can hear the peer voice.

### 3.3 Calling Party and Called Party Cannot Hear Each Other

This topic describes how to troubleshoot a fault that causes failure for the calling party and the called party to hear each other.

### 3.4 Volume Is Low or High

This topic describes how to troubleshoot the low volume or the high volume.

### 3.5 Noise in the Line

This topic describes how to troubleshoot the noise in the line.

### 3.6 Echoes In the Line

This topic describes how to troubleshoot the echoes in the line.

### 3.7 Voice Is Intermittent

This topic describes how to troubleshoot the voice intermittent.

### 3.8 Call Connection Abnormally

This topic describes how to troubleshoot the call connection abnormally.

### 3.9 Sending and Collecting Digits Fail

This topic describes how to troubleshoot a fault that causes failure to send and collect digits.

### 3.10 Number Is Displayed Abnormally

This topic describes a fault that causes number display abnormal.

### 3.11 Voice Prompt Faults

This topic describes how to troubleshoot voice prompt faults.

### 3.12 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

## 3.1 Background

This topic describes call principles.

### 3.1.1 Number Conversion

The number conversion is used for changing the original the calling number to other number which the peer device requires.

The common number conversion and the priority of number conversion is as follows.

#### 1. Long and short numbers

A user has a short number and a long number. The short number is used for intra-office calls(such as 5000). The long number(such as 68906000), which is allocated by the PSTN, is displayed or is used for a PSTN user to directly call a intra-office user.

#### 2. Number mapping

This method is used when irregular numbers are changed. For example, the long number of user 5000 is 68905000 and the mapping number is 83785000.

#### 3. Number change

A user can change calling and called numbers as required. For example, the long number of user 5000 is 68905000 and the mapping number is 83785000. Set outgoing prefix 7 and perform three types of number changes of the calling number. The settings are as follows:

```
//Add a user and set a long number for the user.  
config add sipue eid 5000 authorizationtype noauth  
config add subscriber eid 5000 dn 5000 longdn 68905000  
//Set number mapping for the user.  
config add numbermap innernum 5000 externnum 83785000  
//Add the number change rule index, and change the number to 68905888.  
config add predeal index 1 changetype modify changepos 0 changelen 32 newdn 68905888  
//Perform the switch between long and short numbers, number mapping, and number change.  
config add prefix dn 7 callcategory basic callattribute local cldpredeal no  
officeselectcode 1 clipredeal yes cliindex 1 climap yes uselongcli yes
```

If a user dials a number that starts with 7, the long calling number, that is, 68905000, is displayed on the called phone. The mapping number 83785000 is displayed only when the user does not set the long number. If the user does not set the long number and number mapping, the number after change, that is, 68905888, is displayed.

#### 4. Trunk bearer

This method is used in the scenario where outgoing prefixes are the same, but numbers are different. If a user dials a number that starts with 7+external, the number 68985000 will displayed when outgoing A office and the number 83786000 will displayed when outgoing B office.

## 3.1.2 Prompt Tone

Prompt tones include the dial tone, service prompt tone, and customized ring back tone (CRBT). The prompt tones are saved in the **voice** file. Faults can be located based on prompt tones.



### NOTE

A **voice.zip** package is located on each Media Resource Unit (MRU), Media Resource Server (MRS), or Main Control Unit (MCU).

Pay attention to the following:

- The SoftCo plays dial tones, keypad tones, and prompt tones (for example, "Sorry. The number you dial does not exist. Try later.") for IAD, IP phone, and SIP users.
- If the peer user is called, the device that plays the prompt tone varies.
  - If a common RBT is heard and the local RBT is enabled on the peer device, the peer device plays the RBT. If the local RBT is not enabled on the peer device, the SoftCo plays the RBT.
  - If the SoftCo is connected to the IP Multimedia Subsystem (IMS). Thanks to the IMS do not support to hear the SoftCo prompt tone, the IMS plays the service prompt tone, for example, "Your call has been restricted".

## 3.2 Only One Party Can Hear the Peer Voice

This topic describes how to troubleshoot a fault that occurs when only one party can hear the peer voice.

### Symptom

After a call is set up, only one party can hear the peer voice.

### Possible Causes

- The network is abnormal.
- The terminal is faulty.
- The Real-time Transport Protocol (RTP) port on the router is disabled.
- The Network Address Translation (NAT) is enabled on the network.
- The codec negotiation specified in the signaling is incorrect.
- SDP messages are abnormal.
- RTP packets are abnormal.
- UDP packets are abnormal.

### Procedure

**Step 1** Check whether the network is normal. For details, see [2.5 Network Information Analysis](#).

If the network is faulty, check whether the network cables are connected correctly.

**Step 2** Change the terminal to check whether the fault is rectified.

**Step 3** Check whether the RTP port of the router is enabled.

If the RTP port is not enabled, see router-related document to enable it.

- Step 4** Check whether there are IP addresses on different segments, such as 192.168.10.10 and 10.40.10.10.

If different IP address segments exist, NAT traversal happens. Use the Session Border Controller (SBC) to map IP addresses or establish a Virtual Private Network (VPN).

- Step 5** Use the commissioning record function to record a call through the trunk and locate the fault (peer or local device).



**NOTE**

The commissioning record function does not apply to the SIP trunk or MRS board.

1. Use the Trivial File Transfer Protocol (TFTP) tool to set the directory as the one storing voice files.
2. Run the **debug recordtone by firstcall pid pra out\_toneid 251 recordtype alerting** or **debug recordtone by firstcall pid pra in\_toneid 251 recordtype alerting** command to record an outgoing or incoming call through the trunk.

[Table 3-1](#) describes the command line interface (CLI) parameters.

**Table 3-1** Parameter description

Parameter	Description	Value Range
by	Recording criteria	<ul style="list-style-type: none"> <li>• Firstcall</li> <li>• Caller</li> <li>• Called</li> <li>• Tkc</li> </ul>
pid	Trunk module name, with the recording criteria being <b>firstcall</b>	<ul style="list-style-type: none"> <li>• At0</li> <li>• Pra</li> <li>• R2</li> <li>• Tup</li> <li>• Isup</li> </ul>
number	Calling/Called number, with the recording criteria being <b>caller</b> or <b>called</b>	–
tkno	Recording criteria: <b>Tkc</b>	0–65535
in_toneid	ID of the channel that stores incoming trunk data recording	251
out_toneid	ID of the channel that stores outgoing trunk data recording	251
recordtype	Recording type	<ul style="list-style-type: none"> <li>• FSK</li> <li>• Alerting</li> <li>• Speaking</li> </ul>

 **NOTE**

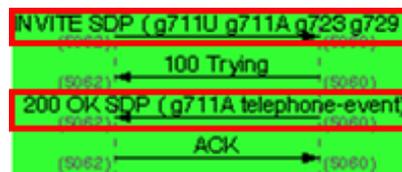
Voice files can be stored only through channel 251. Multiple recording commands cannot be implemented at the same time.

3. (Optional) Manually store voice files that cannot be stored automatically.
  - a. Run the **show record information** command to view **Last record BoardNo** and record the board number.
  - b. Run the **config upload subvoice toneid 251 solt<2-9>** command to save the voice file.
4. Check the quality of the stored voice file to and locate the fault (peer or local device).

**Step 6** Use the packet capture tool to check whether media sets contained in the Session Description Protocol (SDP) messages of the two parties have the same content. The SDP messages of the both parties contain g711A, as shown in [Figure 3-1](#).

If the two parties' SDP messages do not have the same content, change the voice codec type. For details, see [How to Use the Wireshark](#).

**Figure 3-1** Voice codec



To change the voice codec type on the SoftCo, proceed as follows:

1. Run the **show system information** command to view the voice codec type, as shown in [Figure 3-2](#).

**Figure 3-2** Voice codec type on the SoftCo

```
[%SoftCo9500(config)]#show system information

System Information

      Device Type: SoftCo9500
      DomainName: SS
      Net workmode: Single
      IP Address: 10.166.17.94
      SubMask: 255.255.255.0
      MAC Address: 00-c5-3b-8c-73-0b
      Gateway IP Address: 10.166.17.1
      Host IP Address: 10.166.17.218
      CTIServer IP Address: 10.166.17.195
      BillServer IP Setting: 10.166.17.195:2020
      LogServer IP Address: 10.166.17.222
      Voice Code And Decode Type: G711A/G711U/G729/G723
      PertainBoard Type: 8245 HMU 100M
      Master/Slave Flag: Master
      Running Time: 25 Hours 8 Minutes
      Current Time: Date:2010-05-29 Time:11:51:58
      Time Zone: GMT+08:00

==== Command executed success ! ====
```

2. Run the **config system codetype** <g711a&g711u&g729&g723> command to set the voice codec type.
3. Run the **config modify office no**<0-254>**codetype** <g711a&g711u&g729&g723&none> command to change the priority of the voice codec.

To change the voice codec type on the IAD, proceed as follows:

1. Run the **display sip send-capability** command to view the voice codec type.
2. Run the **sip send-capability voip pri**<0-9> <G711a|G711u|G729|G723> **ptime** <10ms|20ms|30ms> command to change the priority of the voice codec.

To change the voice codec type on the IP phone, proceed as follows:

1. Start Microsoft Internet Explorer and enter the IP address of the IP phone (such as http://192.168.104.16) in the address box to go to the IP phone login page.
2. Enter the password (**admin** by default).
3. Click the **ACCOUNT** tab. Set the voice codec type, as shown in Figure 3-3.

**Figure 3-3** Voice codec type on the IP phone

<b>Preferred Vocoder: (in listed order)</b>	choice 1:	PCMA	choice 5:	G.726-32
	choice 2:	PCMU	choice 6:	G.722 (wide band)
	choice 3:	G.723.1	choice 7:	GSM
	choice 4:	G.729A/B	choice 8:	iLBC

**Step 7** Use the packet capture tool to check whether the INVITE SDP message is normal.

1. Check whether the SDP message's media attribute is **sendrecv**, as shown in Figure 3-4. If the media attribute is **sendonly** or **recvonly**, skip to Step 10.

**Figure 3-4** Media attribute

```

Session Initiation Protocol
├─ Status-Line: SIP/2.0 200 OK
├─ Message Header
└─ Message body
    └─ Session Description Protocol
        Session Description Protocol version (v): 0
        └─ Owner/Creator, Session Id (o): 4043 29739 7272939 IN IP4 221.131.228.59
            Session Name (s): -
            └─ Connection Information (c): IN IP4 221.131.228.59
                └─ Time Description, active time (t): 0 0
                    └─ Media Description, name and address (m): audio 35364 RTP/AVP 8
                        └─ Media Attribute (a): sendrecv
                            └─ Media Attribute (a): rtpmap:8 PCMA/8000
                                └─ Media Attribute (a): ptime:20
    
```

2. Check whether the media attribute of the INVITE SDP message sent by the peer device carries the message **silenceSupp:off**, as shown in Figure 3-5.

If **silenceSupp:off** is contained in the INVITE SDP message, run the **config sip silenceofftofax flag off** command to disable **silenceSupp:off** in Config mode.

**Figure 3-5** silenceSupp:off message

```
Message body
  Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): HuaweiSoftCov100R002 12564 12566 IN IP4 10.120.1.146
    Session Name (s): Sip Call
    Connection Information (c): IN IP4 10.120.1.146
    Time Description, active time (t): 0 0
    Media Description, name and address (m): audio 6980 RTP/AVP 8 0
    Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute (a):ptime:20
    Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute (a):ptime:20
    Media Attribute (a): silenceSupp:off - - -
    Media Attribute (a): X-modem
```

**Step 8** Use the packet capture tool to check the RTP packet content.

1. Check whether RTP packets are received and sent bidirectionally.  
If RTP packets are received and sent unidirectionally, skip to [Step 10](#).
2. Run the **ping** <a.b.c.d> command to check whether the SoftCo IP address is used by another device.  
If the IP address is used by another device, the media access control (MAC) address of the RTP packet is incorrect. Run the **config system ip mode static address**<a.b.c.d/M> command in Config mode to change the SoftCo IP address.
3. Use Capsens to convert the RTP message that has been captured to a voice file, and analyze the file. If the voice file is empty, analyze the log of the device that sent the RTP message to locate the fault. For details on how to use Capsens, see [8.1 How to use the Capsens](#).

**NOTE**

Before capturing packets, pay attention to the following points:

- If two broadband terminals, such as two SIP phones, are connected, capture the IP packets on the terminals.
- If the SoftCo is used to connect calls that pass through different trunks, capture the packets on the SoftCo service network port.

**Step 9** Check whether the UDP packet checksum is correct. [Figure 3-6](#) shows the correct checksum.

If the checksum is incorrect, skip to [Step 10](#).

**Figure 3-6** Correct checksum

```
User Datagram Protocol, Src Port: netbios-ns (137), Dst Port: netbios-ns (137)
  Source port: netbios-ns (137)
  Destination port: netbios-ns (137)
  Length: 58
  Checksum: 0x3b63 [correct.]
    [Good Checksum: True]
    [Bad Checksum: False]
```

**Step 10** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.3 Calling Party and Called Party Cannot Hear Each Other

This topic describes how to troubleshoot a fault that causes failure for the calling party and the called party to hear each other.

### Symptom

After a call is set up, neither of the two parties can hear the peer voice.

### Possible Causes

The possible causes are as follows:

- The Field Programmable Gate Array (FPGA) version information is not the same as that described in the release notes delivered with the product.
- The network is abnormal.
- The terminal is faulty.
- The RTP port on the router is disabled.
- The Network Address Translation (NAT) is enabled on the network.
- The codec negotiation in the signaling is incorrect.
- SDP messages are abnormal.
- RTP packet are abnormal.
- UDP packet are abnormal.

### Procedure

- Step 1** Run the **show version** command to check whether the Field Programmable Gate Array (FPGA) version information is the same as *SoftCo Version Configuration Table* in the release notes delivered with the product, as shown in [Figure 3-7](#).

If no, obtain the correct version to upgrade.

**Figure 3-7** Version information

```
[%SoftCo9500(config)]#show version

Version information of slot 1
-----
Pcb          Version : SC12MCU VER -1
Primary BIOS Version : 101
Secondary BIOS Version : 000
Software     Version : V100R002C03
Logic       Version : (U35)255 (U5)001 (U24)302
EHP         Version :

Version information of RBB
-----
Pcb          Version : SC12RBB VER 0
==== Command executed success ! ====
```

- Step 2** Perform steps 1 to 9 in [3.2 Only One Party Can Hear the Peer Voice](#).

**Step 3** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.4 Volume Is Low or High

This topic describes how to troubleshoot the low volume or the high volume.

### Symptom

The volume is low or high in the line.

### Possible Causes

The device gain is improper.

### Procedure

**Step 1** If the AT0 trunk interface is used for interconnecting devices, change the AT0 outgoing gain (**dagain**) and incoming gain (**adgain**) by follow way. The default values of the outgoing and incoming gains are **6**.

Run the following commands to adjust the outgoing and incoming gains:

- If the AT0 trunk interface provided by the SoftCo5816 chassis is used, run the **config modify at0tkc mpu portinboard 0-7 dagain <0-12> adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo5816 board is used, run the **config modify at0tkc card slot <1-4> portinboard 0-7 dagain <0-12> adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo9500 board is used, run the **config modify at0tkc slot <2-9> portinboard 0-7 dagain <0-12> adgain <0-12>** command. The **playdefvol** is play tone gain. The **ipchandefvol** is IP channel gain. The **tdmchandefvol** is TDM channel gain. The default values of the **playdefvol**, the **ipchandefvol** and **tdmchandefvol** gains are **12**.

**Step 2** If another trunk is used to connect devices, run the **config modify mrsparas playdefvol <0-15> ipchandefvol <0-15> tdmchandefvol <0-15>** command to increase the outgoing and incoming gains. In this command, default values of **playdefvol**, **ipchandefvol**, and **tdmchandefvol** are **12**.

**Step 3** If an IAD is used, run the **dsp send-gain <2-64>** and **dsp recv-gain <2-64>** commands to change the outgoing and incoming gains.

**Step 4** If an IP Phone is used, press the arrow keys to adjust the volume.

**Step 5** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.5 Noise in the Line

This topic describes how to troubleshoot the noise in the line.

## Symptom

There is noise in the line.

## Possible Causes

The possible causes are as follows:

- The SoftCo is not grounded.
- The terminal is faulty.
- The device gain is improper.
- The peer device is abnormal.

## Procedure

- Step 1** Check whether the SoftCo is grounded.
- Step 2** Change the terminal to see whether the fault is rectified.
- Step 3** See [3.4 Volume Is Low or High](#) Step 1 to Step 2 to reduce device gain.
- Step 4** If there is cross-talk, check whether there is interference on the line.
- Step 5** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.
- End

## 3.6 Echoes In the Line

This topic describes how to troubleshoot the echoes in the line.

## Symptom

There is echoes in the line.

## Possible Causes

The possible causes are as follows:

- The network is abnormal.
- The echoes are acoustic feedback from the user's microphone.
- The device gain is improper.
- The EC function is not enabled.
- The Echo Cancellation (EC) function is incorrectly.

## Procedure

- Step 1** Check whether the network status is normal. For details, see [2.5 Network Information Analysis](#).
- Step 2** The PSTN user covers the microphone to check whether the echoes disappear.

If yes, the echoes are acoustic. Acoustic echoes generally occur when the hands-free function is enabled. To prevent acoustic echoes, disable the hands-free function.

**Step 3** If a SoftCo is used, run the **show system voip argu algo g711** command to check whether the EC function is **ON**, as shown in [Figure 3-8](#).

If no, run the **config system voip algo g711 echocancel on** command to enable the EC function.

**Figure 3-8** Echo cancellation

```
[%SoftCo5816(config)]#show system voip argu algo g711
System VoIP Arguments
      PacketTime: 20
      EchoCancel: [ON]
      GainControl: 14
      SilenceSuppression: [OFF]
      JitterThreshold(min/max/init): 20(ms)/200(ms)/20(ms)
==== Command executed success ! ====
```

**Step 4** See [3.4 Volume Is Low or High](#) Step 1 to Step 2 to reduce device gain.

**Step 5** Contact the peer office's engineers to check whether the Echo Cancellation (EC) function is correctly configured in the peer gateway.

**Step 6** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.7 Voice Is Intermittent

This topic describes how to troubleshoot the voice intermittent.

### Symptom

The voice is intermittent in the line.

### Possible Causes

The possible causes are as follows:

- The network is abnormal.
- The silence suppression function is enabled.

### Procedure

**Step 1** Check the network status. see [2.5 Network Information Analysis](#).

**Step 2** Use the commissioning record function to record a call through the trunk and locate the fault (peer or local device). For details, see [Step 5](#).

**Step 3** If a SoftCo is used, run the **show system voip argu algo g711** command to check whether the silence suppression function is **OFF**, as shown in [Figure 3-9](#).

If no, run the **config system voip algo g711 silencesup off** command to disable the silence suppression function.

**Figure 3-9** Silence suppression

```
[~SoftCo5816(config)]#show system voip argu algo g711

System VoIP Arguments

                PacketTime: 20
                EchoCancel: [ON]
                GainControl: 14
                SilenceSuppression: [OFF]
JitterThreshold(min/max/init): 20(ms)/200(ms)/20(ms)

==== Command executed success ! ====
```

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.8 Call Connection Abnormally

This topic describes how to troubleshoot the call connection abnormally.

### Symptom

Call connection faults are as follows:

- Connect a Call takes more than ten second.
- The caller will hear busy tone after callee unhooking.

### Possible Causes

The possible causes are as follows:

- The prefix attribution is abnormal.
- If the user is SIP or H.248, the digitmap is incorrect.
- The length of VU collected digits is incorrect.
- The codec negotiation in the signaling is incorrect.

### Procedure

**Step 1** Run the **show prefix dn <string>** command to check the **minlen** whether less than intra-office number.

if yes, run the **config modify prefix dn <string> minlen <string>** command to modify.

**Step 2** If the user is SIP or H.248, run the **show digitmap** command to check digitmap whether meet requirement.

The least length of the default digitmap is four. If the intra-office number less than it, run the **config add digitmap protocol h248 name <string> value <string>** command and **config modify digitmap protocol sip value <string>** command to modify digitmap.

For example, if the intra-office number is 700, run the **config modify digitmap protocol sip value 7XX** command to modify.

**Step 3** Check the length of VU collected digits whether more than extension number. The length of VU collected digits as shown in [Figure 3-10](#) is displayed.

If yes, please modify the length of VU collected digits to intra-office number length. For example, the intra-office number is four, you can change the thirty-two to four.

**Figure 3-10** The length of VU collected digits

```
300  calledNum = ""
301  NumLen = 0
302  PlayCollect 312,312,250,true,32,'#','*','#',#,20,NumLen,calledNum
400  switch(NumLen)
500  {
600      case 1:
602      {
700          if(calledNum == groupNo)
701          {
800              CallTransfer transferNum,1024,1023,1050,1060,2000,1070
801          }
```



**NOTE**

You can obtain the VU script file in the release notes delivered with the product( that is **Script** file).

**Step 4** See [3.2 Only One Party Can Hear the Peer Voice](#) Step 5 to check the device voice codec.

**Step 5** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.9 Sending and Collecting Digits Fail

This topic describes how to troubleshoot a fault that causes failure to send and collect digits.

### Symptom

Sending and collecting digits faults are as follows:

- Two-stage dialing fails.
- The voice prompt "Please input called number." is played all the time.
- The dialed number is incorrect.
- The connected user is not the user that the calling party calls.

### Possible Causes

The possible causes are as follows:

- The digit collection mode does not match that of the peer device. For example, one end uses the RFC2833 digit collection mode but the other end does not.
- The digits that have been sent or received are incorrect.

## Procedure

**Step 1** If the SoftCo uses the SIP trunk to connect to the peer device, check whether the SoftCo and the peer device use the same digit collection mode.

Contact the peer engineer to check the digit collection mode and the value of **PayloadTpye**. If the RFC2833 digit collection mode is enabled on the peer device, perform the following operations on the SoftCo:

1. Run the **show cdsp argu** command to view the digit collection mode. The RFC2833 digit collection mode is disabled by default. This means that **RFC2833Flag:0** is displayed.
2. Run the **config cdsp rfc2833switch <on|off>** command in Config mode to enable the RFC2833 digit collection mode.
3. Run the **show cdsp payloadtype** command to check whether the RFC2198 load (the value of **PayloadTpye**) is the same as that of the peer device.

If the values are different, run the **config cdsp payloadtype rfc2833 value<0-255>** command in Config mode to change the value.

**Step 2** Capture logs by referring to [2.3 Log Analysis](#). Check whether the calling or called number that has been sent or received meets the specific requirement. For example, when an intra-office user initiates a call, the calling number transmitted to the Public Switched Telephone Network (PSTN) must be a long number.

If the calling or called number does not meet the requirement, modify it by referring to **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the Number Change**.

**Step 3** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.10 Number Is Displayed Abnormally

This topic describes a fault that causes number display abnormal.

### Symptom

The number is displayed fault is as follows:

- The number is displayed incorrectly.
- The number cannot be displayed.

### Possible Causes

The possible causes are as follows:

- The Calling Line Identification Presentation (CLIP) service is enabled. This is, when the caller has CLIP service, the callee phone can not display the caller number.
- The number change rule is incorrect.

## Procedure

**Step 1** Run the **show subscriber dn**<string>**type newservice** command to check whether the user has enabled the Calling Line Identification Restriction (CLIR) service. (CLIR is not set by default.)

If yes, run the **config modify subscriber dn** <string> **operatnewservice del newserVICerights clir** command to delete the configuration.

**Step 2** Check whether the number change rule of the outgoing prefix is correct.

1. Run the **show prefix dn**<string> command to check whether the outgoing prefix is configured with the number change rule.

If yes, use the following command to check and modify the number change rule:

- Run the **show subscriber** command to whether the user long number is correct (value of LongDn). If the configuration is incorrect, run the **config modify subscriber dn** <string> **longdn** <string> command to modify.
- Run the **show predeal index** <0-1023> command to check whether the number change rule is correct. If the number change is incorrect, run the **config modify predeal** command to modify.
- Run the **show numbermap** to check whether the number mapping is correct. If the number mapping is incorrect, run the **config modify numbermap num** <string> **newexternnum** <string> to modify.

Figure 3-11 indicates configure the long nuber (The value of UseLongCLI is YES ), the number change (value of CLIPredeal Information) and the number mapping (value of NumberMap Information).

**Figure 3-11** Outgoing prefix number change

```
[%SoftCo5816(config)]#show prefix dn 8
Prefix
SubPBXNo Prefix CallCategory CallAttribute CustomAttribute ChargeIndex SelectOfficeCode
-----
0      8      basic      local      null      ---      0
MinLen MaxLen UseLongCLI If4PSTN WaitNextNumTimerLen PwdCallLimit CallerNumAuth
-----
0      32      YES      YES      0 (ms)      YES      No

CLIPredeal Information
CLIPredeal CLIIndex ChangeType ChangePos ChangeLen NewDn
-----
YES      1      MODIFY      0      8      68918080

NumberMap Information
CLDMap CLIMap
-----
NO      YES

==== Command executed success ! ====
```

2. Run the **show tgld** command to check whether the trunk bearer has been configured. The trunk bearer has been configured, as shown in Figure 3-12.

If yes, skip to Step a.

**Figure 3-12** Trunk bearer

```
[%SoftCo5816(config)]#show tglD
Trunk Group Load
CallSourceNo OfficeNo Prefix PeerRingBack CLIPreDeal CLIIndex CldPreDeal CldInde
x
-----
-
1 0 6890 NO YES 1 NO NULL
TKSeizePoint Protocol
-----
NULL SIP
==== Command executed success ! ====
```

**Step 3** See [2.3 Log Analysis](#) to capture the log information, check whether the caller number and callee number match the peer requirement. For example, the intra-office user is caller user, you must send the long number to the PSTN.

If no, modify it by referring to **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the Number Change**.

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.



**NOTE**

If the AT0 trunk is used for the connection, the calling number is displayed as a POTS number and cannot be changed.

----End

## 3.11 Voice Prompt Faults

This topic describes how to troubleshoot voice prompt faults.

### Symptom

Voice prompt faults are as follows:

- When an intra-office user picks up the phone, no voice prompt can be heard, but the user hears the busy tone after a while.
- When an intra-office user picks up the phone and dials the service number, there is no voice prompt or the voice prompt is incorrect.
- When an outer-office user calls an intra-office user who is not in idle state, no voice prompt can be heard. The intra-office user hears the busy tone after a while.
- When an outer-office user calls an intra-office user who is not in idle state, the voice prompt is incorrect.

### Possible Causes

The possible causes are as follows:

- The service configuration is incorrect.

- The Media Resource Unit (MRU) or Media Resource Server (MRS) on the SoftCo9500 is defective.
- Files in the **voice.zip** package on each board are not synchronized.
- Files in the **voice.zip** package are incorrect.
- The voice codec mode on the terminal does not match that on the SoftCo.
- The peer device does not support the voice prompt function.

## Procedure

**Step 1** If only a service voice prompt is incorrect, check the service configuration. For details about the service configuration, see Services Configuration(CLI).

**Step 2** On the SoftCo9500, run the **show board** command to check whether the MRU and MRS are in **OK** state.

If the MRU and MRS are in **Fault** state, proceed as follows:

1. Log in to the MRU through the serial port and run the **show version** command to check whether the version of the MRU or MRS is the same as that of the MCU. For details on how to log in to the MRU through the serial port, see How to Connect to the SoftCo Through the Serial Port or by Telnet.

If the version of the MRU or MRS is different from that of the MCU, see the *SoftCo Upgrade Guide* to upgrade the version.

2. Remove and reinstall the MRU and MRS, or replace the MRU and MRS.

**Step 3** On the SoftCo9500, run the **config download file voice slot<2-9>** command to synchronize the **voice.zip** package on the MCU to the MRU and MRS.

**Step 4** Run the **config upload file voice** command to download the **voice.zip** file in Config mode. After the package is decompressed, check whether the .pcm file in the **voice** folder is correct.

If the file is incorrect, obtain the **voice** folder from the software package, make the **voice.zip** file, and load it to the SoftCo. For details on how to make the **voice.zip** file, see VU Service.

**Step 5** See Step 5 in [3.2 Only One Party Can Hear the Peer Voice](#) to check whether the IP phone, SoftCo, and IAD use the same codec.

**Step 6** Check whether the peer device supports the voice prompt function on the SoftCo.

If the function is not supported, change the voice prompt to the error code prompt.

1. Run the **show softargu** command to view the software parameter ID (the value of **Type**) of a particular voice prompt.

```
[%SoftCo5816(config)]#show softargu
```

```
All Arguments in SoftArgu Table
```

Type	Value	Meaning
0	--	System max user number
1	--	System max cpu ration
...		
278	--	Way of SIPTK errcode indication[1:183(default) 0:errcode]//SIP trunk
...		
307	--	Msg type for IMS[0:180(forward) 1: 183(Colorring)]//IMS
308	--	Way of PRATK errcode indication[1:183(default) 0:errcode]//PRA trunk

```
...  
310 --          Way of R2TK errcode indication[1:183(default) 0:errcode]//R2 trunk
```

2. Run the **config softargu type<0-512>value 0** command in Config mode to change the voice prompt to the error code prompt (the default value is 1).



#### NOTE

If the ISUP or TUP trunk is used, run the **config protocol isup officeno<0-254>sendabnormaltone <yes|no>** command to change the voice prompt to the error code prompt.

- Step 7 Capture logs by referring to [2.3 Log Analysis](#). Check whether signaling interactions between trunks are correct.

- If the error code or signal releasing is incorrect, check the data configuration.
- If the data configuration is correct, contact engineers of the peer device.

- Step 8 If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 3.12 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

### 3.12.1 IAD User Cannot Be Called

This topic describes how to troubleshoot a fault that occurs when an IAD user is called.

#### Symptom

User numbers on the SoftCo range from 801 to 806. An IAD user uses 801 and POTS users use 802 to 806. All users can call peer device users, but the peer device users cannot dial user number 801.

#### Possible Causes

Run the **show prefix** command to check the prefix information. The minimum number length is 4 and the maximum number length is 24. When a user dials 802#, the number length calculated by the user box is 4. The number length calculated by the IAD, however, is 3, which is smaller than the minimum number length. As a result, the IAD user cannot be called.

#### Procedure

- Run the **config modify prefix dn 8 minlen 3 maxlen 24** command to modify the minimum number length.

----End

### 3.12.2 Connecting a Call from an IAD User to a Peer Device User Takes a Long Time

This topic describes how to troubleshoot a fault that occurs when a call is connected from an IAD user to a peer device user.

## Symptom

When an IAD user calls a peer device user, the call takes up to 10 seconds to be connected.

## Possible Causes

The way of number report on the SoftCo is not the same as the IAD. The software parameter corresponding to a number is enabled on the SoftCo. The IAD reports all numbers at one time.

## Procedure

- Step 1** Run the **show softargu** command to check value of the **Type** field corresponding to **sip support iad no pond switch**.

```
[%SoftCo5816(config)]#show softargu
```

```
All Arguments in SoftArgu Table
```

Type	Value	Meaning
0	--	System max user number
1	--	System max cpu ration
...		
282	0x1	sip support iad no pond switch[1:(open) 0:close default:0]

- Step 2** Run the **config softargu type<0-512>value 0** command to disable the software parameter **sip support iad no pond switch**.

----End

## 3.12.3 A Caller Hears the Voice Prompt That "The number you dialed does not exist" When Calling an Intra-Office User

This topic describes how to troubleshoot a fault that a caller hears the voice prompt that "The number you dialed does not exist" when calling an intra-office user.

## Symptom

Intra-office user A or external user C hears the voice prompt "The number you dialed does not exist" when calling intra-office user B.

## Possible Causes

The intra-office prefix is not configure.

## Procedure

- Run the **config add prefix dn <string> callcategory basic callattribute inter minlen 0 maxlen 8** command to configure the related prefix.

----End

### 3.12.4 No User Can Make a Call

This topic describes how to troubleshoot a fault that no user can make a call.

#### Symptom

No user on the SoftCo can call other intra-office or outer-office users. In addition, no outer-office users can call users on the SoftCo.

#### Possible Causes

Run the **show billpool state** command to check the number of bills is greater than 90% of the capacity. Call restriction on the full bill pool may be enabled. When this function is enabled, all the calls are restricted if the number of bills is greater than 90% of the capacity (SoftCo5816 capacity: 50,000 bills; SoftCo9500 capacity: 240,000 bills).

#### Procedure

- Run the **debug debug\_pid bill\_server cmd 45 p1<0-4294967295>p2<0-4294967295>** command to delete some bills so that the number of bills is less than 80% of the bill pool capacity. In the command, **p1** indicates the start bill to be deleted and **p2** indicates the number of bills to be deleted. You can delete bills multiple times. Command execution takes a long time.

----End

### 3.12.5 UA5000 Users Cannot Call Users Whose Number Prefix Is 05370821

This topic describes how to troubleshoot the fault that UA5000 users cannot call users whose number prefix is 05370821.

#### Symptom

- On the SoftCo, the outgoing prefix is set to 0 and number change is performed (the first digit of the called number is deleted). UA5000 users can call mobile users but cannot call fixed-line users whose number prefix is 05370821.
- On the SoftCo, the outgoing prefix is set to 05370821 and number change is not performed. UA5000 users cannot call fixed-line users whose number prefix is 05370821.

#### Possible Causes

The default H.248 digitmap on the SoftCo does not match the actual dialing scheme.

#### Procedure

**Step 1** Run the **show digitmap** command to check the default H.248 digitmap.

**Step 2** In Config mode, run the **config delete digitmap protocol h248** command to delete the default digit map.

**Step 3** In Config mode, run the **config add digitmap protocol h248 name ua5000 value [FE]xxEx.F|[FE]xxF|ExxExx.Exx.F|ExxExx.Exx.Exx.F|EExx|Exxx|x** command to add a digitmap that matches user's actual dialing requirement.

**----End**

# 4 Trunk Faults

---

## About This Chapter

Use different methods for troubleshooting faults on different trunks.

### [4.1 Background](#)

This topic describes the background on trunks, which helps to troubleshoot faults.

### [4.2 E1 Trunk Exception](#)

This topic describes how to troubleshoot E1 trunk exceptions.

### [4.3 Call Fails on the PRA Trunk](#)

This topic describes how to troubleshoot a call fail that occurs on the PRA trunk.

### [4.4 SIP Trunk Fault](#)

This topic describes common SIP trunk fault and how to troubleshoot the fault occurs on the SIP trunk.

### [4.5 Call Fails on the SS7 Trunk](#)

This topic describes how to troubleshoot a call fails that occurs on the SS7 trunk.

### [4.6 Call Fails on the AT0 Trunk](#)

This topic describes how to troubleshoot a call fails that occurs on the AT0 trunk.

### [4.7 Call Fails on the R2 Trunk](#)

This topic describes how to troubleshoot a call fails that occurs on the R2 trunk.

### [4.8 Typical Cases](#)

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

## 4.1 Background

This topic describes the background on trunks, which helps to troubleshoot faults.

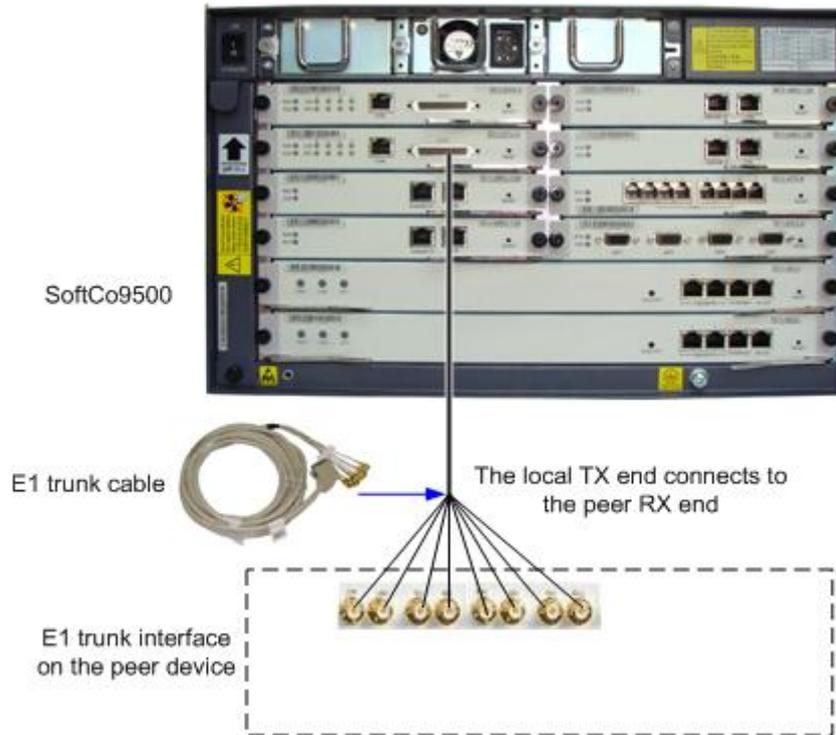
## 4.1.1 E1 Trunk Cable

This topic describes the SoftCo physical connection with the E1 trunk cable and common E1 alarms.

### Physical Connection

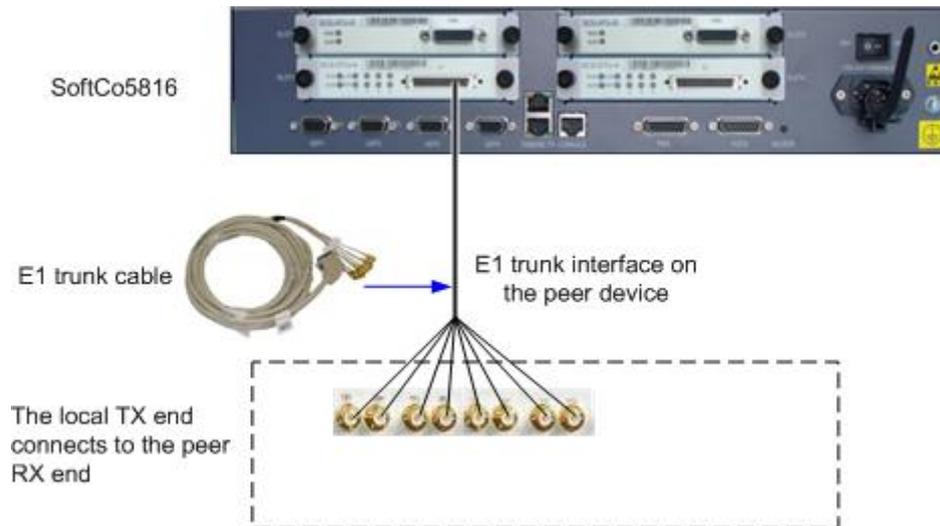
The SoftCo9500 is connected to the peer device with the E1 cable, as shown in [Figure 4-1](#).

**Figure 4-1** Connecting the SoftCo9500 to the peer device with the E1 cable



The SoftCo5816 is connected to the peer device with the E1 cable, as shown in [Figure 4-2](#).

**Figure 4-2** Connecting the SoftCo5816 to the peer device with the E1 cable



**NOTE**

Currently, the following cables are used:

- E1 unbalanced cable with the transmission distance being about 200 m.
- E1 balanced cable with the transmission distance being about 150 m.

## E1 Alarms

E1 alarm types include:

- Fault alarms

A fault alarm is generated when the device is faulty. You must rectify the fault so that the system can be restored. Common fault alarms include:

- Loss of signal (LOS)

When a device cannot detect a signal, an LOS alarm is generated. When the device detects the signal, the LOS alarm is cleared.

- Alarm indication signal (AIS)

The peer device cannot correctly receive frame alignment signals.

- Loss of frame alignment (LFA)

An LFA alarm and an LOS alarm occur simultaneously. The LFA alarm is cleared after frame alignment.

- Remote alarm indication (RAI):

An RAI alarm indicates that the peer device cannot detect any signals. This may occur because the SoftCo fails to send signals, the SoftCo is disconnected, or the peer device fails to receive signals.

- Event alarms

Event alarms do not normally require any handling. Common event alarms include CRC4 error, BER of Frame Alignment > 1e-3, and SLIP.

### 4.1.2 PRA Trunk

The primary rate adaptation (PRA) trunk signaling process varies according to the user type.

## Basic Concept

Integrated Services Digital Network (ISDN) has two type of velocity:

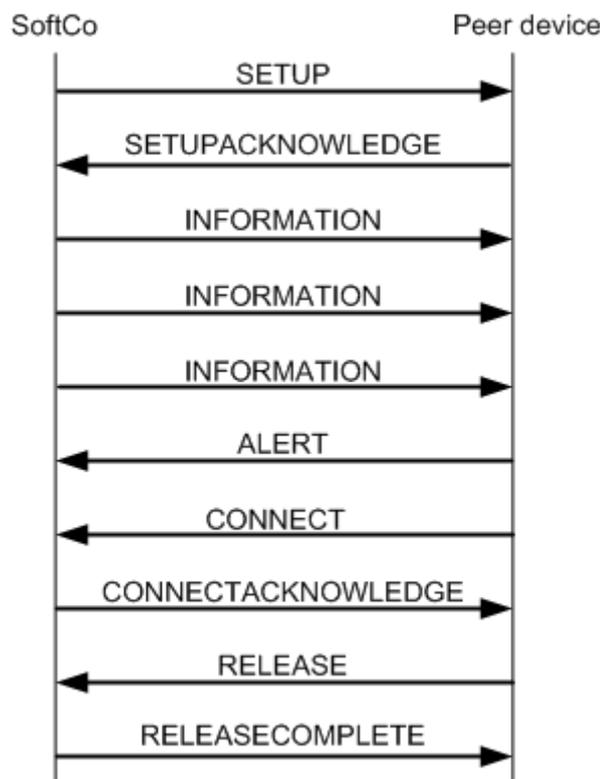
- Basic Rate Interface (BRI)  
It has two B channel and one D channel, this is 2B+D. The velocity of B channel are 64 kbit/s which use to bear sound, image and data communication. The velocity of D channels are 16 kbit/s which uses to indicate the Public Switched Telephone Network how deal with the B channels.
- Primary Rate Interface (PRI)  
It has 30B+D and 23B+D type. Both B channels and D channels are 64 kbit/s digital channels. Except for the United States, all countries use 30 B channels and one D channel.

## Signaling Process

The signaling process varies according to the user type.

- A POTS user in SoftCo calls a user in peer device, as shown in [Figure 4-3](#).

**Figure 4-3** Normal signaling process for a call initiated by a POTS user

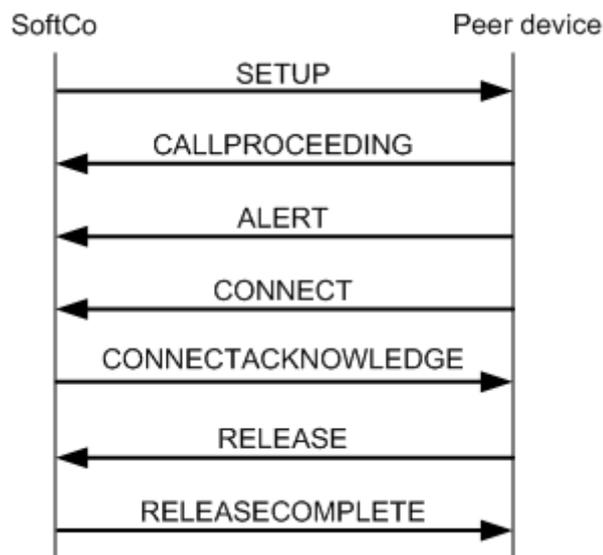


The signaling process is described as follows:

1. When an intra-office user calls an peer user, SoftCo sends a SETUP message to peer device, requesting to set up a call.
2. Peer device sends a SETUPACKNOWLEDGE message, indicating that the call is being set up.

3. SoftCo sends an INFORMATION message with the information required for setting up the call(For example, caller number and callee number).
  4. Peer device sends an ALERT message, indicating that the phone of the peer user rings.
  5. Peer device sends a CONNECT message, indicating that the peer user picks up the phone.
  6. SoftCo sends a CONNECTACKNOWLEDGE message, indicating that the office user A has learned that the peer user has picked up the phone.
  7. The intra-office user starts to talk with the peer user.
  8. Peer device sends a RELEASE message, indicating that the peer user hangs up.
  9. SoftCo sends a RELEASECOMPLETE message, indicating that the intra-office user hangs up. SoftCo and peer device release channel resources.
- A SIP user in SoftCo calls a user in peer device, as shown in [Figure 4-4](#).

**Figure 4-4** Normal signaling process for a call initiated by a SIP user



The signaling process is described as follows:

1. When an intra-office user calls an peer user, SoftCo sends a SETUP message, requesting to set up a call.
2. Peer device returns a CALLPROCEEDING message, indicating that the call is being set up.
3. Peer device sends an ALERT message, indicating that the phone of the peer user rings.
4. Peer device sends a CONNECT message, indicating that the peer user picks up the phone.
5. SoftCo sends a CONNECTACKNOWLEDGE message, indicating that the intra-office has learned that the peer user has picked up the phone.
6. The intra-office user starts to talk with the peer user.
7. Peer device sends a RELEASE message, indicating that the peer user hangs up.
8. SoftCo sends a RELEASECOMPLETE message, indicating that the intra-office user hangs up. SoftCo and peer device release channel resources.



**NOTE**

If the intra-office user is callee user, the signaling process is oppositional.

### 4.1.3 SIP Trunk

This topic describes the basic concept and signaling process for the SIP trunk.

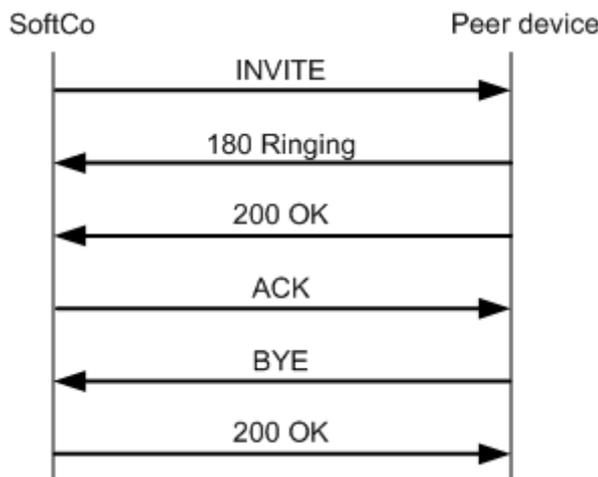
#### Basic Concept

Session Initiation Protocol (SIP) is a signaling control protocol at the application layer and is used to create, modify, and terminate two-party or multi-party sessions. Session Description Protocol (SDP) is used to describe multimedia communication sessions for session announcement and invitation, and multimedia session initialization.

#### Signaling Process

Figure 4-5 shows the normal signaling process.

Figure 4-5 Normal signaling process



The normal signaling process is as follows:

1. When an intra-office user attempts to call an peer user, SoftCo sends an INVITE message to set up a call.
2. Peer device sends a 180 Ringing message indicating that the peer user's phone is ringing.
3. Peer device sends a 200 OK message indicating that the peer user has picked up the phone.
4. SoftCo sends an ACK message indicating that the intra-office user is talking with the peer user.
5. Peer device sends a BYE message indicating that the peer user has hung up.
6. SoftCo sends a 200 OK message indicating that the intra-office user has been informed that the office user B has hung up. The intra-office user then hangs up.

When the SoftCo functions as a device that initiates calls, common error codes are as follows:

- 488: indicates that the call cannot be connected. Check the media type supported by the peer device.

- 486: indicates that the called party is busy. Check whether the terminal of the called party is configured with call transfer or whether the call is forwarded to a phone in busy state.
- 403: indicates that the call is rejected. The peer device does not pass authentication or the called number does not exist.
- 480: indicates a temporary fault. The call may expire.
- 400: indicates that the message format is incorrect. Check whether the message format can be identified by the peer device.
- 500: indicates that the server is faulty. Contact the peer device engineers.

When the SoftCo functions as a device that receives calls, common error codes are as follows:

- 503: indicates that the CPU has reached the threshold and the system limits calls.
- 403: indicates that the trunk may be not configured or there is no call right.

## 4.1.4 SS7 Trunk

The SS7 trunk has different statuses and sustain the TUP and ISUP signaling.

### Trunk Circuit Status

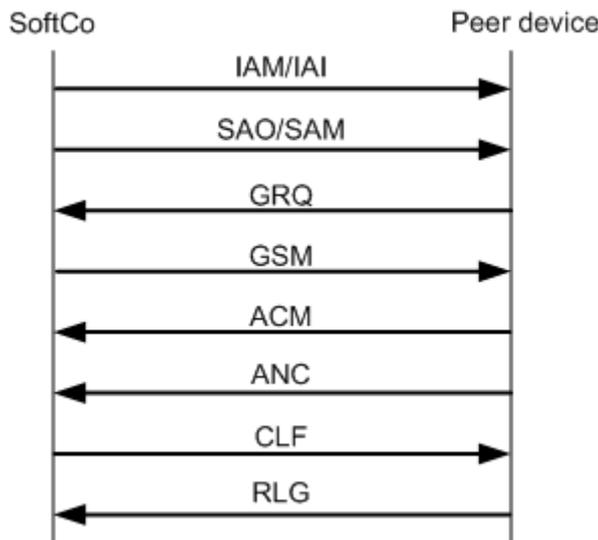
The trunk circuit status is described as follows:

- **IDLE**: Indicates that the circuit is normal.
- **BUSY**: Indicates that the circuit is being used.
- **UNKNOWN**: Indicates that the circuit waits for the response of RSC and GRA messages.
- **FAULT**: Indicates that the circuit is faulty because of incorrect configuration.
- **UNINSTALL**: Indicates that the circuit is not configured.
- **AvailBlock**: Indicates that the circuit can initiate calls only.
- **UnavailBlock**: Indicates that the circuit can receive calls only.

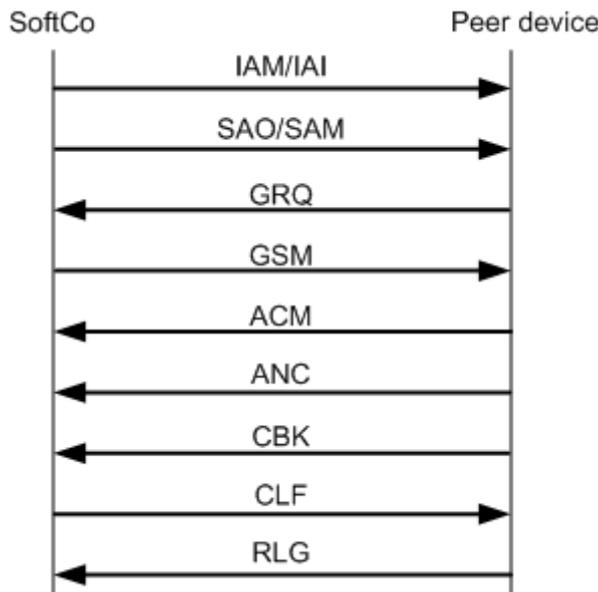
### TUP signaling process

- [Figure 4-6](#) and [Figure 4-7](#) show the normal signaling process.

**Figure 4-6** Normal TUP signaling process (intra-office user first hangs up)



**Figure 4-7** Normal TUP signaling process (peer user first hangs up)

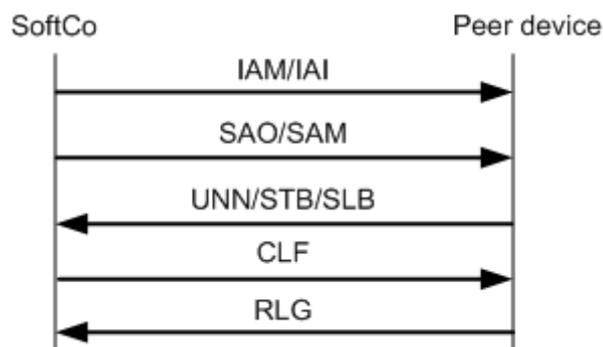


The normal signaling process is as follows:

1. When intra-office user calls peer user, SoftCo device sends an IAM message or an IAI message containing the information about initial address.
2. SoftCo device sends an SAO message or an SAM message containing the information about peer user address.
3. If other information is required for connecting the call, peer device sends a GRQ message requesting other information.
4. After receiving the GRQ message, SoftCo device encapsulates the required information into a GSM message. Then the GSM message is sent. In this case, all the information required for establishing the call is collected.

5. Peer device locates the user according to the information and sends an ACM message. Intra-office user plays the RBT and peer user phone rings.
  6. After peer user picks up the phone, SoftCo device sends an ANC message indicating that peer user has picked up the phone.
  7. Intra-office user starts to talk with peer user.
  8. The mode of sending messages according to the hangup sequence.
    - If intra-office user first hangs up, SoftCo device sends a CLF message indicating that intra-office user has hung up.
    - If peer user first hangs up, peer device sends a CBK message indicating that peer user has hung up. Then intra-office user hangs up and sends a CLF message.
  9. Peer device sends an RLG message indicating that peer device is informed that intra-office user has hung up.
- [Figure 4-8](#) shows the call failure signaling process.

**Figure 4-8** TUP call failure signaling process



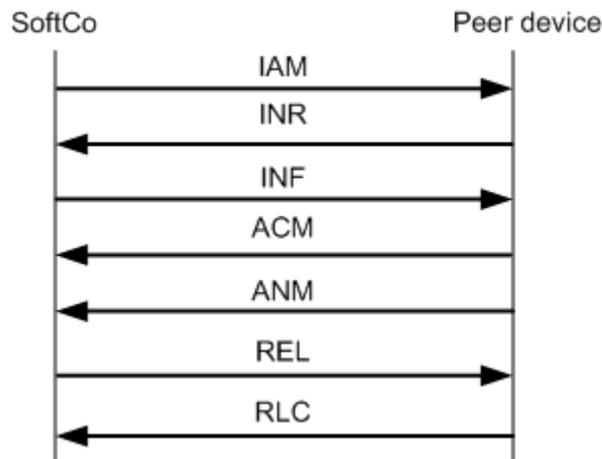
When intra-office user fails to call peer user, peer device sends a message according to the user status.

- If peer user number does not exist, peer device sends a UNN message to SoftCo device.
- If peer user is busy, peer device sends an STB message or an SLB message to SoftCo device.

## ISUP signaling process

- [Figure 4-9](#) shows the normal ISUP signaling process.

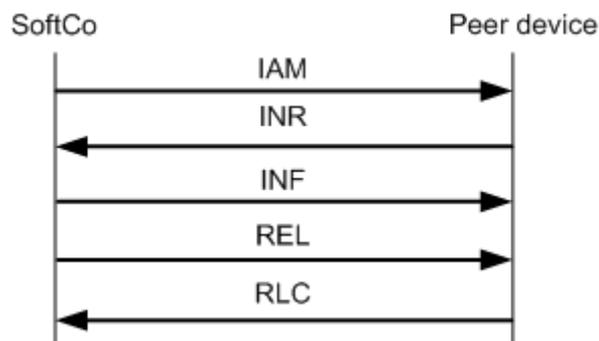
**Figure 4-9** Normal ISUP signaling process



The normal ISUP signaling process is as follows:

1. When intra-office user calls peer user, intra-office users sends an IAM message containing the information about peer user address.
  2. Peer device returns an INR message containing other information required for establishing the call.
  3. SoftCo device sends an INF message containing the information required for establishing the call.
  4. Peer user device locates the user according to the information and sends an ACM message. intra-office user plays the RBT and peer user phone rings.
  5. Peer user sends an ANM message indicating that peer user has hung up.
  6. Intra-office user starts to talk with peer user.
  7. If intra-office user first hangs up, SoftCo device sends an REL message indicating that intra-office user has hung up.
  8. peer device device sends an RLC message indicating that peer device is informed that intra-office user has hung up.
- [Figure 4-10](#) shows the call failure signaling process.

**Figure 4-10** ISUP call failure signaling process



When intra-office user fails to call peer user, peer device sends an REL message. The REL message contains call failure causes.

## 4.1.5 AT0 Trunk

This topic describes the basic concept of the AT0 trunk and incoming and outgoing calls through the AT0 trunk.

The AT0 analog trunk is also called the loop trunk. With the AT0 trunk, the peer line can be used as the trunk line of the local device and the local device functions as the analog user. The number of peer lines determines the number of local trunks. The AT0 trunk uses DTMF signals and pulse signals (**AT0\_DTMF** and **AT0\_PULSE**) to transmit numbers. The maximum distance is 3 to 5 km.

AT0 trunks are classified into outgoing trunks, incoming trunks, and bidirectional trunks.

- AT0 outgoing trunk: A local user calls a peer user through the AT0 trunk. In this case, the AT0 trunk functions as the signal transmission channel.
- AT0 incoming trunk: A peer user calls a local user through the AT0 trunk. In this case, the AT0 trunk receives and parses some analog signals and reports the results.

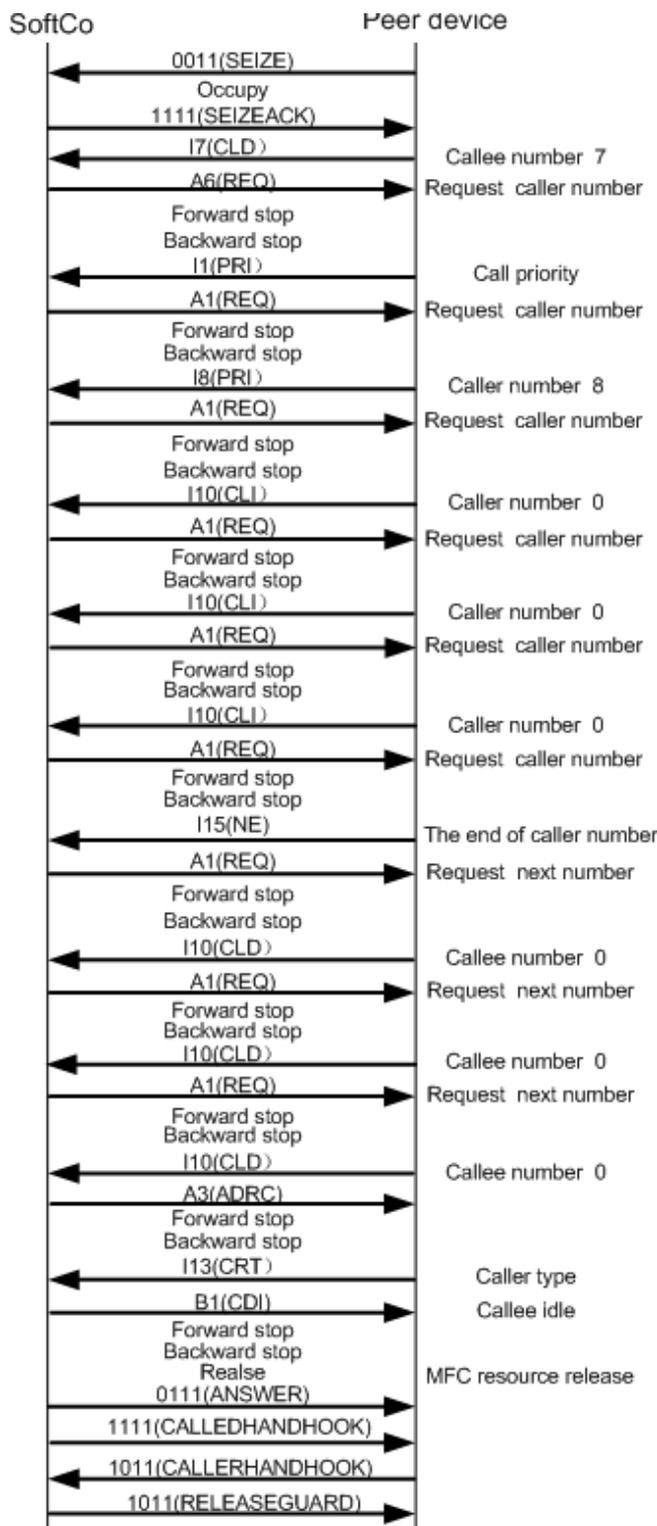
## 4.1.6 R2 Trunk

Channel associated signaling (CAS) indicates that the signaling and voice are transmitted in the same channel. R2 signaling is the CAS.

### Signaling Process

An peer user (number 8000) calls an intra-office user (number 7000). [Figure 4-11](#) shows the normal signaling process.

**Figure 4-11** Normal signaling process



The following describes the signaling process:

1. An peer user calls an intra-office user. peer device sends a seizing signal.
2. SoftCo returns a seizing-acknowledgement signal.

3. SoftCo requests the calling and called numbers from peer device, and then peer device sends the numbers to SoftCo.
4. SoftCo sends status information about the intra-office user to peer device.
5. If the peer user is available, SoftCo sends a response.
6. The intra-office user talks with the peer user.
7. The intra-office user hangs up. SoftCo sends a CALLEDHANDHOOK message.
8. The peer user hangs up. peer device sends a CALLERHANDHOOK message. The talk ends.

## 4.2 E1 Trunk Exception

This topic describes how to troubleshoot E1 trunk exceptions.

### Symptom

E1 trunk faults are as follows:

- An LOS indicator light is on.
- A frame slip alarm or an error code alarm is generated during signaling tracing with the Operation and Maintenance Unit (OMU).

### Possible Causes

The possible causes are as follows:

- The SoftCo is not grounded.
- The clock sources at both ends are asynchronous.
- The E1 trunk cable has failed.
- The board has failed.
- The peer device has failed.

### Procedure

**Step 1** Ensure that the SoftCo is grounded.

**Step 2** Run the **show system clocksource information** command to check whether the clock source is configured.

If the clock source is not configured, as shown in [Figure 4-12](#), run the configuration command.

- SoftCo9500: **config system clocksource primaryslot<2-9> primarye1<0-3>**
- SoftCo5816: **config system clocksource primaryslot<1-4> primarye1<0-3>**

**Figure 4-12** Clock source not configured

```
[%SoftCo5816(config)]#show system clocksource information

System Clock Source Information
Clocksource is not config yet!

System Clock Source:    [Oscillator]

==== Command executed success ! ====
```

- Step 3** Check whether the cable is correctly connected, and whether the cable is excessively long or damaged.
1. Check whether the TX port of the SoftCo is connected to the RX port of the peer device.
  2. Check whether the cable is connected correctly or whether the connector is rusty.  
Reconnect or replace the E1 cable to check whether the fault has been rectified.
  3. Check whether the cable is excessively long according to the cable specifications.  
Usually, the E1 cable must be shorter than 150 m. The exact cable length limitation depends on the cable specifications.
- Step 4** Connect the TX port of the SoftCo to the RX port of the peer device, and check whether the LOS indicator on the board is off.  
If the indicator is still on, replace the board.
- Step 5** Contact the peer office's engineer to check whether the peer device is running normally.
- Step 6** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.
- End

## 4.3 Call Fails on the PRA Trunk

This topic describes how to troubleshoot a call fail that occurs on the PRA trunk.

### Symptom

The following faults often occur on the PRA trunk:

- A busy tone is heard when an incomplete called number is dialed.
- A busy tone is heard when a complete called number is dialed.

### Possible Causes

The possible causes are as follows:

- The E1 trunk cable is faulty or connected incorrectly.
- The clock source is not configured.
- The intra-office and outgoing prefixes are not configured.
- The board status is abnormal.
- The trunk circuit is faulty.

- The network positions (values of **Position**) do not match at both ends.
- The CRC is enabled on one end, but is disabled on the other end.
- The number which is sent by SoftCo do not meet the requirement of peer device.
- Protocol versions do not match. North America and European standards are available for PRA trunks; Q.931 is available for Layer 3 signaling protocol.

## Procedure

**Step 1** Check that the E1 trunk cable is normal and that the clock source is configured. For details, see [4.2 E1 Trunk Exception](#).

**Step 2** Run the **show prefix callcategory basic** command to check whether the intra-office and outgoing prefixes are configured.

- If no, run the **config add prefix** command to configure.

```
[%SoftCo9500(config)]#show prefix callcategory basic
```

```
Prefix
```

```
SubPBXNo Prefix CallAttribute OfficeSelectCode RegGroupId
```

```
-----
```

```
==== Command executed success ! ====
```

- If prefixes are configured, check that there are no conflicting prefixes.

**Step 3** Run the **show board** command on the SoftCo9500 or the **show card** command on the SoftCo5816 to check whether the DTU and MRU are in **OK** state.

If the DTU and MRU are in **FAULT** state, proceed as follows to rectify the fault:

1. Log in to the DTU or MRU through a serial port and run the **show version** command to check whether the version of the DTU or MRU is the same as that of the MCU. For details on how to log in to the DTU or MRU through a serial port, see [How to Connect to the SoftCo Through the Serial Port or by Telnet](#).

If the versions are different, see *SoftCo Upgrade Guide* to upgrade the DTU or MRU.

2. Remove and reinstall, or replace the DTU and MRU.

**Step 4** Run the **show tkc office no<0-254>** command to check whether the trunk circuit status is **IDLE** (value of **State**).

- If the circuit is in **ISOLATE** state, run the **config cancelisolate board slot n<2-9>** command on the SoftCo9500 or the **config cancelisolate card slot<1-4>** command on the SoftCo5816 to cancel board isolation.
- If the circuit is in **FAULT** state, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the PRA Trunk Interconnection Data** to check whether the trunk data is correct.

**Step 5** Run the **show pralink** command to check whether the trunk link status is **OK** (value of **State**), which indicates that the link status is normal.

If the value of **State** is **FAULT**, check link configuration.

- The network position (value of **Position**) of one end must match that of the peer device. That is, when one end is the network side, the other end must be the user side. The SoftCo is configured as the user side.

- The timeslot (value of **TS**) of PRA links and the timeslot of the peer device must both be 16.

If the configuration is not correct, run the **config modify pralink linkno <0-255> slot n trunkport n position <user|network> ts <0-31>** command to modify.

**Step 6** Check whether the RFA indicator on a board is on.

If yes, check whether the CRC is enabled on the peer device. If no, run either of the following commands to enable the CRC on the SoftCo:

- For the SoftCo9500, run **config modify board slot<2-9> type dtu trunkport<0-3> officeno<0-254> crc4flag yes**.
- For the SoftCo5816, run **config modify card slot <1-4> type dtu trunkport<0-3> officeno<0-254> crc4flag yes**.

**Step 7** See [2.4 Signaling Analysis](#) trace the signaling to check the call fail reason and deal with the fault.

1. When intra-office calls external user, check whether the SoftCo sends a SETUP message to the peer device, as shown in [Figure 4-13](#).
  - If no, check whether the PRA trunk configuration is correct.
  - If yes, check the content of the SETUP message. Verify that the calling number sent from the SoftCo to the peer device meets the requirements of the peer device.

Set the value of **uselongcli** to **yes** (displaying the long number of the calling party on the phone of the called party) or change the calling number. See **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the Number Change** to configure the data.

**Figure 4-13** SETUP message

```

PRA Signal: Direction: OUTGOING; IE number: 5; signal is whole? Yes
Protocol discriminator: 0x08
Length of call reference value: 2
Call reference value: 0x0247
Message type: SETUP
IE Name: Sending complete.
  IE Name: Bearer Capability; IE Length: 3
  IE Name: Channel identification; IE Length: 3
  IE Name: Calling party number; IE Length: 5
    Numbering plan identification: Unknown(0)
    Type of number: Subscriber number(4)
    Presentation indicator: Presentation allowed(0)
    Screening indicator: User-provided, not screened(0)
    Number digits: 131 caller number
  IE Name: Called party number; IE Length: 9
    Type of number: Subscriber number(4)
    Numbering plan identification: Unknown(0)
    Number digits: 92312266 callee number
    
```

2. When intra-office calls external user, check whether the calling/called number type and numbering plan in the SETUP message meet the requirements of the peer device. The **Type of number** indicates the number type and the **Numbering plan identification** indicates the numbering plan in the [Figure 4-13](#). If no, proceed as follows to modify the configuration:
  - Use software parameters to modify the numbering plan.
    - a. Run the **show softargu** command to check the value of **Tp** field corresponding to **PRA NumberPlan Identification**.

```
[%SoftCo5816(config)]#show softargu
```

All Arguments in SoftArgu Table

Type	Value	Meaning
0	--	System max user number
1	--	System max cpu ration
...		
257	--	PRA NumberPlan Identification

- b. Run the **config softargu type<0-512>value<value>** command to modify the numbering plan. The **type** parameter indicates the software parameter ID. The values of **<value>** are as follows:

- 0: unknown numbering plan
- 1: ISDN telephony numbering plan
- 3: data numbering plan
- 4: telegraph numbering plan
- 8: national standard numbering plan
- 9: private numbering plan
- 14: false calling line numbering plan
- 15: reserved for expansion

- Run the **config protocol pra officeno<0-254>typeofcldnumber<value1>typeofclinumber<value2>** command to configure the number type. The **value1** parameter indicates the called number type; the **value2** parameter indicates the calling number type. The types are as follows:

- unknown: unknown number
- international: international number
- national: national number
- networkspecific: network-specific number
- subscriber: subscriber number
- in: intelligent number
- abbreviated: abbreviated number
- reservedforextension: reserved for expansion
- default: default type

3. When external user calls intra-office, check whether the format of the SETUP message sent by the peer device is correct and whether the SETUP message carries the called number information. If no, contact the peer device engineers.

**Step 8** Contact the peer engineer to check whether the protocol version of peer device match the SoftCo.

**Step 9** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.4 SIP Trunk Fault

This topic describes common SIP trunk fault and how to troubleshoot the fault occurs on the SIP trunk.

## 4.4.1 Call Fails on the SIP Trunk

This topic describes how to troubleshoot a call fails that occurs on the SIP trunk.

### Symptom

The following faults often occur on the SIP trunk:

- A busy tone is heard when an incomplete called number is dialed.
- A busy tone is heard when a complete called number is dialed.

### Possible Causes

The possible causes are as follows:

- The intra-office and outgoing prefixes are not configured.
- The trunk data is not configured.

### Procedure

**Step 1** Check that the intra-office and outgoing prefixes are configured. For details, see Step 2 in [4.3 Call Fails on the PRA Trunk](#).

**Step 2** Run the **show protocol sip** command to check whether the trunk data is configured.

If no, run the **config protocol sip officeno<0-254> domainname <string> peerport <0-6000> maxcallnum <1-240>** command to configure the trunk data.

```
[%SoftCo9500(config)]#show protocol sip
```

```
SIP
```

```
Office IPAddress DomainName LocalUdpPort LocalTcpPort PeerPort TransportType MaxCallNum
-----
DefaultCLDNum1 DefaultCLDNum2 CallerPhoneContext CalleePhoneContext DeviceType OTMN
-----
SupportInfo CRBT SupportCliPredeal ImsConnId
-----
```

```
==== Command executed success ! ====
```

**Step 3** See [2.4 Signaling Analysis](#) to trace signaling process, check the call fail reason and deal with the fault.

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.4.2 Call Interrupts on the SIP Trunk

This topic describes how to troubleshoot a call interrupts that occurs on the SIP trunk.

### Symptom

During The SoftCo user call the peer device user, the call is interrupted suddenly.

## Possible Causes

The possible causes are as follows:

- Multiple SIP terminals are registered with the same account.
- A heartbeat connection has not been established.
- The peer device is abnormal.

## Procedure

**Step 1** Check whether multiple SIP terminals are registered with the same account.

If yes, quit the other SIP terminals.

**Step 2** Check whether the peer device supports the session heartbeat function.

If no, run the **config modify sip parameter usesessiontimer no** command to disable the session heartbeat function.

**Step 3** Contact the peer office's engineer to check whether the peer device is running normally.

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.4.3 Fail to Call an IMS User Through the SIP Trunk

An IMS user cannot be called through the SIP trunk.

### Symptom

When calling the IMS user, the intra-office user hears the busy tone.

### Possible Causes

The possible causes are as follows:

- The registration group configuration is incorrect.
- The calling number format does not meet IMS (86+area code+number) requirements.
- The registration group ID is not set.

### Procedure

**Step 1** Run the **show reggroup** command to check whether the value of **Status** is **OK**.

If no, check whether the **IMPI** field, **IMPU** field and **Password** field match the configuration of IMS.

```
[%SoftCo9500(config)]#show reggroup
```

```
Register group
```

```
Id  IMPI                               IMPU                               Password Expires(*s)
---  -
-----
1   +862083785000@guangdong.test.com +862083785000@guangdong.test.com 111      300
```

```

2 +862083785001@guangdong.test.com +862083785001@guangdong.test.com 111 300
Status
-----
FAULT
FAULT

----- Matched record number : [9] -----
    
```

**Step 2** Run the **show prefix** command to check whether the calling number change configuration of the prefix meets IMS requirements.

If no, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the Number Change** to configure the data.

**Step 3** Run the **show subscriber** command to check whether the registration group ID (that is, **RegGroupId**) corresponding to the user is set. The registration group ID is set, as shown in [Figure 4-14](#).

If no, run the **config modify subscriber dn <string> reggroupid <0-1600>** command to change the value of **RegGroupId**.

**Figure 4-14** User registration group

```

[%SoftCo5816 (config)]#show subscriber

Subscriber

GlobalPortNo DomainName TermName Dn LongDn SubPEXNo State BindUserNum PriorLevel
-----
168 5001 5001 5001 0 FAULT NA Default
169 5002 5002 5002 0 FAULT NA Default
170 5003 5003 5003 0 FAULT NA Default
RegGroupId RemarkInfo DirectDialTEC PotswmiType CallerTimeLimit CalledTimeLimit
-----
0 - -- --- ---
1 - -- --- ---
0 - -- --- ---

----- Matched record number : [3] -----

==== Command executed success ! ====
    
```

 **NOTE**

If users on other trunks connected to the SoftCo call IMS users, run the **show office user** command to check whether the value of **RegGroupId** is set.

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.4.4 IMS Users Cannot Be Invited to Join a Conference Through the SIP Trunk

IMS users cannot be invited to join a conference through the SIP trunk.

## Symptom

The SoftCo fails to invite IMS users to join a conference.

## Possible Causes

The possible causes are as follows:

- The registration group password or account is incorrect.
- The calling number format does not meet IMS (86+area code+number) requirements.
- The registration group ID is not added to the outgoing calling number of the conference.

## Procedure

**Step 1** Use the **show reggroup** command to check whether the value of **Status** is **OK**.

If no, check whether the **IMPI** field, **IMPU** field and **Password** field match the configuration of IMS.

```
[%SoftCo9500(config)]#show reggroup

Register group

Id IMPI                               IMPU                               Password Expires(*s)
-- -----
-----
1 +862083785000@guangdong.test.com +862083785000@guangdong.test.com 111      300
2 +862083785001@guangdong.test.com +862083785001@guangdong.test.com 111      300
Status
-----
FAULT
FAULT

----- Matched record number : [9] -----
```

**Step 2** Run the **show prefix** command to check whether the calling number change configuration of the prefix meets IMS requirements.

If no, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the Number Change** to configure the data.

**Step 3** Use the **show subpbx** command to check whether the outgoing calling number (**OutConfDn**) and the registration group ID (**RegGroupID**) of the conference are set. The outgoing calling number and registration group ID are set, as shown in [Figure 4-15](#).

If no, run the **config modify subpbx no <0-251> outconfdn <string> reggroupid <0-1600>** command to modify the values of **OutConfDn** and **RegGroupID**.

**Figure 4-15** User registration group

```
[%SoftCo5816 (config)]#show subpbx

SubPbx info

NO. SubPBX Name   MaxTkcCount UsedTkcCount OfficeSelectCode MaxConfRoom UsedConfRoom
-----
0  defaultsubpbx N/A         0           N/A             N/A         0
MaxConfSeat UsedConfSeat InterConfDn OutConfDn   RegGroupID  ifUseCRBT
-----
N/A         0           000          862068988000 1           NO

==== Command executed success ! ====
```

**Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.5 Call Fails on the SS7 Trunk

This topic describes how to troubleshoot a call fails that occurs on the SS7 trunk.

### Symptom

The following faults often occur on the SS7 trunk:

- A busy tone is heard when an incomplete called number is dialed.
- A busy tone is heard when a complete called number is dialed.

### Possible Causes

The possible causes are as follows:

- The E1 trunk cable is faulty or connected incorrectly.
- The clock source is not configured.
- The intra-office and outgoing prefixes are not configured.
- The board status is abnormal.
- The trunk circuit is faulty.
- The trunk link status is faulty.

### Procedure

**Step 1** Check that the E1 trunk cable is normal and that the clock source is configured. For details, see [4.2 E1 Trunk Exception](#).

**Step 2** Check whether the intra-office and outgoing prefixes are configured and whether the board status is normal. For details, see step 2 and step 3 in [4.3 Call Fails on the PRA Trunk](#).

**Step 3** Check whether the trunk circuit is faulty.

Run the **show tkc office no<0-254>** command to check whether check whether the trunk circuit status is **IDLE** (value of **State**).

- If the circuit is in **FAULT** state, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the ISUP/TUP Trunk Interconnection Data** to check whether the trunk data is correct.
- If the circuit is in **UNINSTALLED** state, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the ISUP/TUP Trunk Interconnection Data** to configure.

**Step 4** Run the **show mtplink** command to check whether the trunk link status is **OK** (value of **State**).

If no, reset or replace the board to troubleshoot the fault.

**Step 5** See [2.4 Signaling Analysis](#) to trace signaling process, check the call fail reason and deal with the fault.

**Step 6** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 4.6 Call Fails on the AT0 Trunk

This topic describes how to troubleshoot a call fails that occurs on the AT0 trunk.

### Symptom

The following faults often occur on the AT0 trunk:

- A (SoftCo user) is unable to call B (peer device user).
- B is unable to call A.

### Possible Causes

The possible causes are as follows:

- The intra-office and outgoing prefixes are not configured.
- The board status is abnormal.
- The SoftCo is not grounded.
- The FXS port is abnormal.
- The AT0 trunk circuit is faulty.
- The AT0 trunk line is abnormal.

### Procedure

**Step 1** Check whether the intra-office and outgoing prefixes are configured and whether the board status is normal. For details, see step 2 and step 4 in [4.3 Call Fails on the PRA Trunk](#).

**Step 2** Check whether the SoftCo is grounded.

**Step 3** Check whether indicators on the ATU are normal. For details about indicators, see [2.2 Alarm Analysis](#).

**Step 4** Check whether the FXS port connected to the ATU is normal.

Disconnect the ATU from the FXS port and connect the POTS phone to the FXS port to check whether the call can be connected. If no, replace the FXS port.

**Step 5** Check whether the AT0 trunk circuit is faulty.

Run the **show tkc slot<2-9>** command on the SoftCo9500 or the **show tkc slot<1-4>** command on the SoftCo5816 to check the trunk circuit status.

- If the circuit is in **ISOLATE** state, run the **config cancelisolate board slot<2-9>** command on the SoftCo9500 or the **config cancelisolate card slot<1-4>** command on the SoftCo5816 to cancel board isolation.
- If the circuit is in **FAULT** state, go to [Step 6](#).

**Step 6** Check whether the AT0 trunk line is normal.

Connect the phone to the AT0 trunk line and call users of the peer device. If calls to multiple users cannot be connected, contact the peer device engineers.

**Step 7** Check whether the ATU is faulty by using the OMU. If yes, check whether the ATU is installed.



**NOTE**

You can obtain the OMU from <http://support.huawei.com>. For details on how to use the OMU, see *OMU Client Help*.

**Step 8** If the fault persists, please obtain [1.6 Technical Support](#).

----End

## 4.7 Call Fails on the R2 Trunk

This topic describes how to troubleshoot a call fails that occurs on the R2 trunk.

### Symptom

The following faults often occur on the R2 trunk:

- A busy tone is heard when an incomplete called number is dialed.
- A busy tone is heard when a complete called number is dialed.

### Possible Causes

The possible causes are as follows:

- The E1 trunk cable is faulty or connected incorrectly.
- The clock source is not configured.
- The intra-office and outgoing prefixes are not configured.
- The board status is abnormal.
- The trunk circuit is faulty.
- The register type is not MFC.
- The R2 adaptation script is not imported.

## Procedure

- Step 1** Check that the E1 trunk cable is normal and that the clock source is configured. For details, see [4.2 E1 Trunk Exception](#).
- Step 2** Check whether the intra-office and outgoing prefixes are configured and whether the board status is normal. For details, see step 2 and step 3 in [4.3 Call Fails on the PRA Trunk](#).
- Step 3** Run the **show tkc office no**<0-254> command to check whether the trunk circuit status is **IDLE** (value of **State**).
- If the circuit is in **ISOLATE** state, run the **config cancelisolate board slot**<2-9> command on the SoftCo9500 or the **config cancelisolate card slot**<1-4> command on the SoftCo5816 to cancel board isolation.
  - If the circuit is in **FAULT** state, see **SoftCo Product Document > Installation and Commissioning > Configure Voice Service (Command Line Mode) > Configuring Trunk Data > Configuring the R2 Trunk Interconnection Data** to check whether the trunk data is correct.
- Step 4** Run the **show cassignal** command to check whether the register type is MFC(value of RegisterType).
- If no, run the **config modify cassignal casname** <string> **registertype mfc**command to modify the register type.
- Step 5** Check whether the R2 adaptation script has been imported.
- Run the following command to check whether the R2 adaptation script has been imported:
- ```
show line send id <0-254>
show line report id <0-254>
show register send id <0-254>
show register report id <0-254>
```
- Check whether the number of displayed lines is the same as that of command lines corresponding to the script. If no, delete the configuration and re-import the adaptation script. To delete the configuration, run the following commands:
- ```
config delete line send id <0-254>
config delete line report id <0-254>
config delete register send id <0-254>
config delete register report id <0-254>
```
- Step 6** See [2.4 Signaling Analysis](#) to trace signaling process, check the call fail reason and deal with the fault.
- Step 7** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.
- End

## 4.8 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

## 4.8.1 Clock Source Configuration Fault

If the clock source is incorrectly configured, exceptions will occur, for example, noise and frame slip.

### Symptom

SoftCo A is connected to SoftCo B through the Primary Rate Adaptation (PRA) trunk. There are noises during a call or when you dial \*125 to query a number. A frame slip alarm is reported when you use the Operation and Maintenance Unit (OMU) to analyze alarms.

### Possible Causes

SoftCo A synchronizes time to SoftCo B and SoftCo B synchronizes time to SoftCo A, resulting in a failure to synchronize the clock sources.

### Procedure

- Step 1** If SoftCo A is connected to the carrier's network through the PRA trunk, change the SoftCo A clock source, and use the carrier's time.

Proceed as follows to modify the clock source:

- Run the **config system clocksource primaryslot<2-9>primarye1<0-3>** command in Config mode on the SoftCo9500.
- Run the **config system clocksource primaryslot<1-4>primarye1<0-3>** command in Config mode on the SoftCo5816.

- Step 2** If SoftCo A is connected to the carrier's network through the Session Initiation Protocol (SIP) trunk, run the **config system clocksource oscillator** command in Config mode to change the SoftCo A clock source to the default time.

#### NOTE

The IMS is connected to SoftCo through the SIP trunk. The Online Charging System (OCS) is connected to the SoftCo through the SIP trunk, and the SoftCo functions as an intermediate device. The SoftCo clock source uses the default time.

----End

## 4.8.2 Calling a PSTN User Through the PRA Trunk Fails

This topic describes how to troubleshoot a fault occurring when a PSTN user is called through the PRA trunk.

### Symptom

The SoftCo is connected to the PSTN through the PRA trunk. The SoftCo boards and the PRA trunk link are running properly. A PSTN user can call an intra-office user, but the intra-office user cannot call the PSTN user.

### Possible Causes

Use the UCMaint to analyze the logs. The values of **Numbering plan identification** and **Number digits** in the SETUP message when the intra-office user is the calling party are found

to be different from those when the PSTN user is the calling party, as shown in [Figure 4-16](#) and [Figure 4-17](#).

**Figure 4-16** Values displayed when the PSTN user functions as the calling party

```

    [-] PRA Signal: Direction: INCOMING; IE number: 7; signal is whole? Yes
        Protocol discriminator: 0x08
        Length of call reference value: 2
        Call reference value: 0x3bfb
        Message type: SETUP
        IE Name: Sending complete;
    [+] IE Name: Bearer Capability; IE Length: 3
    [+] IE Name: Channel identification; IE Length: 3
    [+] IE Name: Progress indicator; IE Length: 2
    [+] IE Name: Display; IE Length: 13
    [-] IE Name: Calling party number; IE Length: 2
        Numbering plan identification: ISDN/telephony numbering plan (Recommendation E.164 [19]) (1)
        Type of number: Unknown(0)
        Presentation indicator: Presentation allowed(0)
        Screening indicator: Network provided(3)
        Number digits:
    [-] IE Name: Called party number; IE Length: 4
        Type of number: Unknown(0)
        Numbering plan identification: ISDN/telephony numbering plan (Recommendation E.164 [19]) (1)
        Number digits: 700
    
```

**Figure 4-17** Values displayed when the SoftCo user functions as the calling party

```

    [-] PRA Signal: Direction: OUTGOING; IE number: 5; signal is whole? Yes
        Protocol discriminator: 0x08
        Length of call reference value: 2
        Call reference value: 0x0247
        Message type: SETUP
        IE Name: Sending complete;
    [+] IE Name: Bearer Capability; IE Length: 3
    [+] IE Name: Channel identification; IE Length: 3
    [-] IE Name: Calling party number; IE Length: 5
        Numbering plan identification: Unknown(0)
        Type of number: Subscriber number(4)
        Presentation indicator: Presentation allowed(0)
        Screening indicator: User-provided, not screened(0)
        Number digits: 131
    [-] IE Name: Called party number; IE Length: 9
        Type of number: Subscriber number(4)
        Numbering plan identification: Unknown(0)
        Number digits: 92312266
    
```

## Procedure

- Step 1** Run the **show prefix dn 9** command to check the outgoing prefix attributes. As shown in [Figure 4-18](#), the calling number change attribute is not configured for the outgoing prefix and the value of **UseLongCLI** is **NO**.

**Figure 4-18** Outgoing prefix attributes

```
[%SoftCo5816(config)]#show prefix dn 9
Prefix
SubPBXNo Prefix CallCategory CallAttribute CustomAttribute ChargeIndex SelectOfficeCode
-----
0          9      basic          local          null          ---          1
MinLen MaxLen UseLongCLI If4PSTN WaitNextNumTimerLen PwdCallLimit CallerNumAuth
-----
0        32      NO          YES          0 (ms)          YES          No
CLDPredeal Information
CLDPredeal CLDIIndex ChangeType ChangePos ChangeLen NewDn
-----
YES         22      DELETE     0          1          -
```

- Step 2** Run the **config add predeal index 5 chanetype modify changepos 0 changelen 24 newdn 700** command to configure the number index.
- Step 3** Run the **config modify prefix 9 clipredeal yes cliindex 5 uselongcli yes** command to modify the outgoing prefix attributes so that the calling number meets PSTN requirements.
- Step 4** Run the **show softargu** command to check the **Type** field corresponding to **PRA NumberPlan Identification**. The software parameter ID corresponding to **PRA NumberPlan Identification** is 257.
- Step 5** Run the **config softargu type 257 value 1** and **config protocol pra officeno 0 typeofclnumber unknown typeofclinumber national** commands to modify the numbering scheme.
- End

### 4.8.3 Prompts for the Extension Number or a User Hears the Busy Tone When the User Makes a Call to the Main Number Through the AT0 Trunk

This topic describes how to troubleshoot a fault that occurs when a user makes a call to the main number through the AT0 trunk.

#### Symptom

The SoftCo is connected to the peer device through the AT0 trunk. The peer device user dials the main number, the system prompts for the extension number. The user dials the extension number, but the system continues to prompt for the extension number.

#### Possible Causes

The SoftCo fails to detect signals because the AT0 line is too long.

#### Procedure

- Increase the AT0 incoming gain (**adgain**). The default AT0 incoming gain is **6**.

- If the AT0 trunk interface provided by the SoftCo5816 chassis is used, run the **config modify at0tkc mpu portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo5816 board is used, run the **config modify at0tkc card slot <1-4> portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo9500 board is used, run the **config modify at0tkc slot <2-9> portinboard 0-7 adgain <0-12>** command.

----End

## 4.8.4 Connecting a Call on the AT0 Trunk Takes a Long Time

This topic describes how to troubleshoot a fault occurring when a call is being connected on the AT0 trunk.

### Symptom

The SoftCo is connected to the peer device through the AT0 trunk. A intra-office user calls a peer device user. The intra-office user hears the ring back tone (RBT) after a long delay (10 seconds or more).

### Possible Causes

The intra-office user use the direct-dialing-in(DDI) service. Therefore, connecting a call takes a long time. You are advised to change the dialing mode to two-stage dialing.

### Procedure

- Run the **config modify office selectcode no <0-254> send2nddialtone yes** command to change the dialing mode to two-stage dialing.

----End

## 4.8.5 Calling Number Is Not Displayed for an AT0 Trunk Call

This topic describes how to troubleshoot a fault occurring when a call is made through the AT0 trunk.

### Symptom

The SoftCo is connected to the PSTN through the AT0 trunk. When a PSTN user calls a SoftCo user, the calling number is not displayed.

### Possible Causes

The possible causes are as follows:

- The calling line identification presentation (CLIP) service has not been enabled for the SoftCo user.
- The SoftCo fails to detect signals because the AT0 line is too long.
- The peer device sends calling numbers using dual tone multi-frequency (DTMF) signals, but the SoftCo supports only frequency shift keying (FSK) signals.

## Procedure

**Step 1** Check whether the CLIP service is enabled for the SoftCo user. If the CLIP service is not enabled, ask the user to subscribe to the CLIP service.

Connect the phone to the AT0 trunk interface provided by the carrier to check whether the calling number is displayed. If yes, the CLIP service is enabled.

**Step 2** Check whether FSK signals are used.

- If the calling number is displayed when an incoming call is received, it indicates that DTMF signals are used. In this case, contact peer office engineers.
- If the calling number is displayed between the first ring and the second ring, it indicates that FSK signals are used.

**Step 3** Increase the AT0 incoming gain (**adgain**). The default AT0 incoming gain is **6**.

- If the AT0 trunk interface provided by the SoftCo5816 chassis is used, run the **config modify at0tkc mpu portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo5816 board is used, run the **config modify at0tkc card slot <1-4> portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo9500 board is used, run the **config modify at0tkc slot <2-9> portinboard 0-7 adgain <0-12>** command.

**Step 4** If the fault persists, contact Huawei customer service center or the local office.

----End

## 4.8.6 Called Party Cannot Hear the Busy Tone After the Calling Party Hangs Up on an AT0 Trunk Call

This topic describes how to troubleshoot a fault occurring when the calling party hangs up on an AT0 trunk call.

### Symptom

The SoftCo is connected to the PSTN through the AT0 trunk. Assume that a PSTN user initiates a call to a SoftCo user. After the PSTN user hangs up, the SoftCo user cannot hear the busy tone.

### Possible Causes

The possible causes are as follows:

- Configuration on the carrier side is incorrect. As a result, the system fails to send busy tone signals after the calling party hangs up.
- The SoftCo fails to detect signals because the AT0 line is too long.

### Procedure

**Step 1** Connect the phone to the AT0 trunk interface provided by the carrier and call the SoftCo user using the PSTN number. Hang up after the call is connected. Then check whether the called party hears the busy tone. If no, contact peer office engineers.

**Step 2** Increase the AT0 incoming gain (**adgain**). The default AT0 incoming gain is **6**.

- If the AT0 trunk interface provided by the SoftCo5816 chassis is used, run the **config modify at0tkc mpu portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo5816 board is used, run the **config modify at0tkc card slot <1-4> portinboard 0-7 adgain <0-12>** command.
- If the AT0 trunk interface provided by the SoftCo9500 board is used, run the **config modify at0tkc slot <2-9> portinboard 0-7 adgain <0-12>** command.

**Step 3** If the fault persists, contact Huawei customer service center or the local office.

----End

## 4.8.7 Call Fails When the Mobile User Calls the Intra-office Analog User on the AT0 Trunk

This topic describes how to troubleshoot a call fails when the mobile user calls the intra-office analog user on the AT0 trunk.

### Symptom

The SoftCo is connected to the carrier through the AT0 trunk. When the mobile user calls the intra-office analog, the incoming call displays 000. When the mobile user hangs up, the intra-office analog rings all the time.

### Possible Causes

The AT0 line is interrupted the environment, then the SoftCo can not detect the analog signal.

### Procedure

**Step 1** Check whether the SoftCo is grounded.

Please remove and reinstall the PGND cables to deal with the fault.

**Step 2** Connect analog user cable of the carrier or the ATU board on the SoftCo to the grounding terminal, which used to clear the environment interruption.

[Figure 4-19](#) indicate the SoftCo uses the PGND cables to connect the analog user cable to the grounding terminal.

**Figure 4-19** Grounding connection



**Step 3** If the fault persists, contact Huawei customer service center or Huawei local offices.

----End

## 4.8.8 Calls Made Through the AT0 Trunk Are Always Connected to the Same Number

This topic describes how to troubleshoot the following fault: When a user calls any SoftCo number, the same phone rings.

### Symptom

The SoftCo is connected to the peer device through the AT0 trunk. When a user calls any SoftCo number, the same phone rings.

### Possible Causes

Run **show protocol at0 officeno <0-254>** command discovering the **defaultclidnum1** field has value. The trunk is configured with the default incoming number.

### Procedure

- Run the **config protocol at0 officeno <0-254> defaultclidnum1 0000** command to delete the default AT0 incoming number.

----End

## 4.8.9 Fault Occurs When a Moderator Invites a User to Join a Conference Call Through the AT0 Trunk

This topic describes how to troubleshoot a fault that occurs when a moderator invites a user to join a conference call through the AT0 trunk.

### Symptom

The SoftCo is connected to the peer device through the AT0 trunk. When a moderator invites a user to join a conference call, only part of the called number is displayed on the conference management page because the dialing speed is low.

### Possible Causes

By default, the SoftCo uses a timer to specify the time for the AT0 module to send a Connect message. If the time set on the timer runs out when a SoftCo user calls a peer device user, the AT0 module sends a Connect message to the calling device regardless of whether the dialing is complete. The remaining digits are sent through voice, and they are not recorded by the SoftCo. Therefore, if the moderator dials slowly, the number in the Connect message may be incomplete.

### Procedure

- Run the **config at0 sendremotefhook condition pole** command to change the control mode.



#### NOTE

The fault can be rectified if the peer device supports polarity reversal signals. If the peer device does not support polarity reversal signals, users must dial a number consecutively.

----End

## 4.8.10 Simultaneous/Sequential Ringing Service Cannot Be Used When a User Number of the Peer Device Connected to the AT0 Trunk Is Used for the Service

This topic describes how to troubleshoot a fault that occurs when a user number of the peer device connected to the AT0 trunk is used for the simultaneous ringing service or sequential ringing service.

### Symptom

The SoftCo is connected to the peer device through the AT0 trunk. When a peer device user number is used for the simultaneous ringing service or sequential ringing service, the simultaneous ringing service or sequential ringing service cannot be used.

### Possible Causes

When a user makes an outgoing call through the AT0 trunk, the SoftCo starts the timer by default during the dialing. When the time set on the timer runs out, the AT0 trunk sends a Connect message to the calling device regardless of whether the dialing is complete. If the simultaneous ringing service or sequential ringing service is enabled, the system assumes that the called party picked up the phone, resulting in the fault.

## Procedure

- Run the **config at0 sendremotefhook condition pole** command to change the control mode.



### **NOTE**

The fault can be rectified if the peer device supports polarity reversal signals. If the peer device does not support polarity reversal signals, users must dial a number consecutively.

**----End**

# 5 Service faults

---

## About This Chapter

The troubleshooting method varies according to service category.

### 5.1 Background

This topic describes the background on supplementary service, which helps to troubleshoot faults.

### 5.2 Call Forwarding Faults

This topic describes how to troubleshoot call forwarding faults.

### 5.3 Transfer Service Faults

This topic describes how to troubleshoot call transfer service faults.

### 5.4 ONLY Service Faults

This topic describes the common One Number Link You (ONLY) service fault and how to troubleshoot the One Number Link You (ONLY) service faults.

### 5.5 Main Number Service Fault

This topic describes the common main number service and how to troubleshoot the main number service fault.

### 5.6 Fax service fault

This topic describes the common main number service and how to troubleshoot the fax service fault.

### 5.7 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

## 5.1 Background

This topic describes the background on supplementary service, which helps to troubleshoot faults.

## 5.1.1 Call Forwarding/Transfer Service

Before using forwarding services, pay attention to priorities of forwarding services.

If a call forwarding service is enabled, all calls are automatically forwarded to a preset number regardless of the called party status.

When using call forwarding services, pay attention to the following points:

- Call Forwarding Unconditional (CFU) service: The CFU service conflicts with the following services: Do-Not-Disturb (DND) service, alarm clock service, absent user service, instant hotline service, call waiting service, and call back on busy (CBB) service. The CFU service takes precedence over the Call Forwarding No Reply (CFNR) service and the Call Forwarding Busy (CFB) service.
- CFNR service: The CFNR service conflicts with the following services: DND service, alarm clock service, absent user service, instant hotline service, and CBB service. The priority of the CFNR service is lower than that of the call waiting service.
- CFB service: The CFB service conflicts with the following services: DND service, alarm clock service, absent user service, instant hotline service, call waiting service, and CBB service.
- Call Forwarding Offline (CFO) service: The priority of the CFO service is lower than that of the DND service.

If the call transfer service is enabled, a user can transfer a call to a third party by pressing the hookflash key during a conversation. The user then automatically quits the conversation. By default, SoftCo users have been assigned the right to transfer calls.

## 5.1.2 ONLY Service

If you use ONLY service, please pay attention to conflict other service.

In the one number link you (ONLY) service, multiple terminals share the same number and supplementary services. The shared number is the number of an intra-office terminal, and is called the ONLY number. The intra-office terminal is the primary terminal and can be specified by the subscriber. When other subscribers dial the ONLY number of the subscriber, the terminals of the subscriber ring according to the preset rules, and the subscriber can select a terminal to answer the phone.

The number of the primary terminal has the ONLY service right. Otherwise, the administrator needs to configure the ONLY service right for the number. In addition, the administrator needs to bind one or more other terminals to the number of the primary terminal so that the number can be shared by multiple terminals.



### NOTE

- The priority is : Service of Call Forwarding Unconditional > Services of Call on Busy > Sequential Ringing Service > ONLY Service > Service of Call Forwarding No Reply. You should delete other service which conflict with ONLY service when you give priority to ONLY service.
- If the last user of sequential ringing refuse the call, the caller can heard the prompt voice.

## 5.1.3 Fax

This topic describes the basic problem about fax.

- Identification

Determining whether a fax machine is a sender or a recipient depends on whether there are papers in the paper tray.

- If there are papers in the paper tray, the fax machine is the sender. If there is no paper in the paper tray, the fax machine is the recipient.
- If there are papers in the paper trays of both machines, both machines consider themselves as the sender, causing fax failures.
- If there is no paper in either paper tray, both fax machines consider themselves as the recipient, causing fax failures.
- Fax mode selection  
Usually, fax machines support the following modes: FAX/TEL, FAX ONLY, and MANUAL. Select a mode as required.
  - FAX/TEL: applicable to the situation that the fax machine is sometimes used as a phone.  
In this mode, if a fax arrives, the fax machine does not ring but directly receives the fax. If a call arrives, the fax machine rings to notify users of the call.
  - FAX ONLY: applicable to the situation that the fax machine is used only as a fax machine.  
In this mode, the fax machine switch calls into faxes and automatically receives all faxes.
  - MANUAL: applicable to the situation that the fax machine is frequently used as a phone.  
In this mode, the fax machine rings for every call. To send or receive faxes, you must enable the fax function manually after receiving a call.
- Memory clearing  
Clear the memory periodically or print the memory data. If the recipient fax machine does not print the received data immediately, the data is stored in the memory. When the memory is used up, the fax machine can no longer receive faxes.
- Ink box selection  
To ensure the print definition, use a high-quality ink box.

## 5.2 Call Forwarding Faults

This topic describes how to troubleshoot call forwarding faults.

### Symptom

Common call forwarding service faults are as follows:

- A call initiated by an intra-office user cannot be forwarded to another intra-office user.
- A call initiated by an intra-office user cannot be forwarded to an external user.

### Possible Causes

If a call made by A to B fails to be forwarded to C, the possible causes are as follows:

- A does not have the right to call C.
- The forwarding number of B are incorrect.
- The call forwarding service of B conflicts with any other service.
- The number used by A does not meet the peer device requirements.

## Procedure

**Step 1** Access the command tree configuration page, and enter the Config mode. For details, see `depg/depg/softco_depg_00149.xml`.

For details about how to use the LMT to run commands, see `depg/depg/softco_depg_00170.xml`.

**Step 2** Run the **show subscriber by dn** *<string>* command to check whether A has the right to call C.

```
[%SoftCo95005500(config)]#show subscriber dn 5001
```

Subscriber

```
Dn LongDn DomainName TermName UserType UserPriorLevel State Source UserState
-----
5001 5001 5001 NORMAL Default FAULT 0 NORMAL
SubPBXNo FaxMode RegGroupId BindUserNum RemarkInfo DirectDialTKC RingTypeIndex
-----
0 BYPASS 0 NA - -
CentrexGroup PickupGroup PbxGroup OutgoingRight IncomingRight Telephonist
-----
- - - INTER/LOCAL INTER/LOCAL/DDD/IDD Yes
AreaNumber IfUseCRBT
-----
NO

UserName faxmailnumber SubState NewMsgs OldMsgs voipdomain BillingType SuspendedFlag
-----
- Invalid 0 0 DefaultDomain Periodic No
PotswmiType
-----
--
```

- If no, run the **config modify subscriber by dn** *<string>* **operateout add outgoingrights** *<inter|local|ddd|idd>* command to add the right.
- If yes, go to the next step. The user number 5001 has intra-office call right and local call right, as shown in above.

**Step 3** If C is a peer device user, run the **show subscriber by dn** *<string>* **type newservice** command to check whether the forwarded-to number for B has been added.

```
[%SoftCo95005500(config)]#show subscriber dn 5001 type newservice
```

New Service

```
Rights Register Information
-----
NORMAL Y
CALL_WAITING
WAKEUP_CALL
OPERATOR DISCONNECT Y
OPERATOR INSERT Y
ABBREVIATIVEDIAL
CLIP Y
CALLLIMIT
```

```
DDS
ABSENT_USER
CFU          Y          [Forward:81378888888#]
CFB
CFNR
CFO
HOOKING_TRANSFER  Y
AMEND_PASSWD    Y          [CalloutPassword:1234]
CALLPARK
```

----- Matched record number : [17] -----

- If no, run the **config activeservice dn <string> type cfu value <string>** command to add the forwarded-to number.
- If yes, go to the next step.

**Step 4** Run the **show subscriber by dn <string> type all** command to check whether the call forwarding service of B conflicts with any other service (see Service Conflict List).

- If yes, modify the configuration as required to avoid the conflicts.
- If no, go to the next step.

**Step 5** Check whether the number of A meets the peer device requirements.

For example, the number of A needs to configure long number when the SoftCo connect to the PSTN.

- If no, see Setting Number Conversion Mode to modify the number change rule.
- If yes, go to the next step.

**Step 6** See Step 4 in [3.10 Number Is Displayed Abnormally](#) to check whether the number change rule is correct.

**Step 7** Run the **show sip convert23tosharp switch** command to check whether the SIP character conversion function is enabled. Value **1** indicates yes and **0** indicates no.

- If the command result is **1**, run the **config sip convert23tosharp flag off** to disable the function.
- If the command result is **0**, go to the next step.

**Step 8** If the fault still persists, collect the fault information and send it to Huawei technical support engineers. For the contact information, see [1.6 Technical Support](#).

**Step 9** If you have performed the preceding steps, make a call to check whether the fault is rectified.

If the call made by A to B can be forwarded to C successfully, the fault is rectified.

----End

## 5.3 Transfer Service Faults

This topic describes how to troubleshoot call transfer service faults.

### Symptom

Common call transfer service faults are as follows:

- Integrated access device (IAD) users fail to press the hookflash key.
- Two-stage dialing fails after users press the hookflash key.

## Possible Causes

If B fails to transfer an ongoing call with A to C by pressing the hookflash key, the possible causes are as follows:

- Two-stage dialing signals fail to be detected due to poor network transmission performance.
- B does not have the right to call C.
- The hookflash function is disabled for B.
- B uses an incorrect call transfer method.
- The hookflash duration on B's phone differs from that on the SoftCo or the hookflash key is unavailable.
- The IAD does not support the hookflash function.
- The digit collection mode set on the SoftCo is different from that set on the IAD.

## Procedure

**Step 1** Check whether the network connection is normal. See [2.5 Network Information Analysis](#).

**Step 2** Run the **show subscriber dn <string>** command to check whether user B has the right to call user C. The user number 5001 has intra-office call right and local call right, as shown in [Figure 5-1](#).

If no, run the **config modify subscriber dn <string> operateout add outgoingrights <inter|local|ddd|idd>** command to assign B the outgoing call right.

**Figure 5-1** Outgoing call right

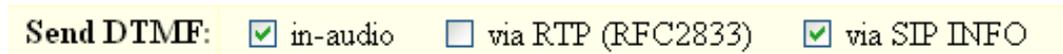
```
[%SoftCo5816(config)]#show subscriber dn 5001
Subscriber
-----
Dn      LongDn  DomainName  TermName  UserType  UserPriorLevel  State      Source  UserState
-----
5001    5001    5001        5001     NORMAL   Default         FAULT     0       NORMAL
SubPBXNo FaxMode  RegGroupId  BindUserNum RemarkInfo DirectDialTKC RingTypeIndex
-----
0       BYPASS  0           NA        -         -               -         -
CentrexGroup OutgoingRight IncomingRight Telephonist AreaNumber IfUseCRBT
-----
-       INTER/LOCAL INTER/LOCAL/DDD/IDD No         NO
-----
UserName  faxmailnumber  SubState  NewMsgs  OldMsgs  voipdomain  BillingType  SuspendedFlag
-----
-         -              Invalid  0        0        DefaultDomain Periodic    No
PotswmiType
-----
---
```

**Step 3** See related IP phone documents to whether the operation and configuration of IP phone is correct.

1. Check whether the transfer service operation performed on the SIP phone is correct.
2. Check whether the phone supports the hookflash function.
3. Check whether the SIP phone configuration is correct.

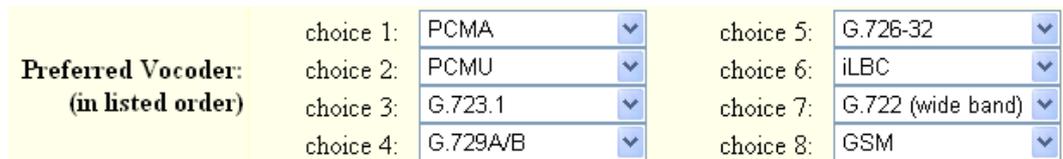
- Check whether the digit collection mode is correct. See [Figure 5-2](#).

**Figure 5-2** Digit collection mode



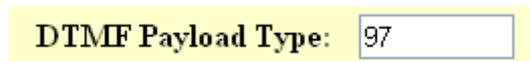
- Check whether the voice codec mode match the SoftCo. as shown in [Figure 5-3](#).

**Figure 5-3** Voice codec mode



- If the RFC2833 digit collection mode is used, check whether the DTMF payload type match the peer device, as shown in [Figure 5-4](#).

**Figure 5-4** DTMF payload type



**Step 4** If B uses a POTS phone, check whether the hookflash duration on the POTS phone is different from that on the SoftCo and check whether the hookflash key is available.

If the hookflash key is unavailable, replace the phone. If the hookflash key functions properly, change the hookflash threshold on the SoftCo by running the corresponding command.

- If the POTS phone is connected to the SoftCo5816 chassis, run the **config hookflashtime mpu lowerlimit <30-1000> upperlimit <30-1000>** command.
- If the POTS phone is connected to the SoftCo5816 user box, run the **config hookflashtime subbox no <1-4>lowerlimit<30-1000>upperlimit<30-1000>** command.
- If the POTS phone is connected to the SoftCo9500 user box, run the **config hookflashtime subbox slot <2-9> port <1-4> lowerlimit <30-1000> upperlimit <30-1000>** command.

**Step 5** If B is connected to the IAD, run **display version** command to check whether the IAD supports the hookflash function.

If no, see IAD-related document to upgrade.

- For IAD101H, IAD102H, IAD104H, support the hookflash before TS0402 V300R001C01.
- For IAD132E(T), support the hookflash before TS3202 V300R001C01.

**Step 6** If the SoftCo uses the SIP trunk to connect to the peer device, check whether the SoftCo and the peer device use the same digit collection mode.

Contact the peer engineer to check the digit collection mode and the value of **PayloadTpye**. If the RFC2833 digit collection mode is enabled on the peer device, perform the following operations on the SoftCo:

1. Run the **show cdsp argu** command to view the digit collection mode. The RFC2833 digit collection mode is disabled by default. This means that **RFC2833Flag:0** is displayed.
2. Run the **config cdsp rfc2833switch <on|off>** command in Config mode to enable the RFC2833 digit collection mode.
3. Run the **show cdsp payloadtype** command to check whether the RFC2198 load (the value of **PayloadTpye**) is the same as that of the peer device.

If the values are different, run the **config cdsp payloadtype rfc2833 value<0-255>** command in Config mode to change the value.

**Step 7** See [2.3 Log Analysis](#) to capture the log and rectify the fault.

**Step 8** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.4 ONLY Service Faults

This topic describes the common One Number Link You (ONLY) service fault and how to troubleshoot the One Number Link You (ONLY) service faults.

### 5.4.1 ONLY Number cannot be bound

This topic describes how to troubleshoot the One Number Link You (ONLY) service faults.

#### Symptom

Numbers cannot be bound by using the ONLY service.

#### Possible Causes

The possible causes are as follows:

- The ONLY service is disabled for the user.
- The number to be bound has the ONLY service.
- The number to be bound (intra-office user) does not exist.
- The ONLY service is enabled for the number to be bound (intra-office user) or the number has been bound to another ONLY number.

#### Procedure

**Step 1** Run the **show subscriber dn <string> type newservice** command to check whether the primary terminal or the number to be bound has ONLY service, as shown in [Figure 5-5](#).

If the primary terminal has no ONLY service, run the **config modify subscriber dn <string> operatnewservice add newservicerights unicall** command to enable the service for the user.

If the number to be bound has ONLY service, change to other number.

**Figure 5-5** ONLY service

```
[%SoftCo5816(config)]#show subscriber dn 5001 type newservice

New Service

Rights          Register Information
-----
NORMAL          Y
CALL_WAITING
WAKEUP_CALL
UNICALL
ABBREVIATIVEDIAL
CLIP            Y
CALLLIMIT
DDS
ABSENT_USER
CFU
CFB
CFNR
CFO
HOOKING_TRANSFER Y
AMEND_PASSWD    Y          [CalloutPassword:1234]
CALLPARK

----- Matched record number : [16] -----

==== Command executed success ! ====
```

- Step 2** Run **show subscriber** command to check whether the number to be bound exists.
- Step 3** Run **show subscriber** command to check whether the ONLY service is enabled for the number to be bound (value of **BindUserNum**).
- Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.4.2 The User Hears the Busy Tone When Dials the ONLY Number

This topic describes how to troubleshoot the uses hears the busy tone when dials the ONLY number.

### Symptom

If a user dials the ONLY number when a bound user is busy, the user hears the busy tone.

### Possible Causes

If a user (A) dials the ONLY number when a bound user (B) is busy, user A hears the busy tone. The possible causes are as follows:

- The **BusySelectMode** field is set to **active** for the ONLY service.
- The main terminal of the ONLY service is not a SIP terminal.

## Procedure

**Step 1** Run the **show subscriber unicall dn** <string> command to check whether the value of **BusySelectMode** is **active**.

If yes, run the **config modify subscriber unicall dn** <string> **busyselectmode all** command to change the value of **BusySelectMode**.

**Step 2** Check whether the main terminal is the SIP terminal.

If no, use a SIP terminal.

**Step 3** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.4.3 The Sequential Ringing Service or the Simultaneous Ringing Service Fails through ONLY Service

This topic describes how to troubleshoot the sequential ringing service or the simultaneous ringing service fails.

### Symptom

The PSTN user cannot use the sequential ringing service or the simultaneous ringing service through ONLY service.

### Possible Causes

The possible causes are as follows:

- The outgoing prefix is not added to the PSTN user number to be bound.
- The interval for sequential ringing is too long.
- A validity period is set for the binding.
- Services conflict.

## Procedure

**Step 1** Run the **show subscriber unicall dn** <string> command to check whether related attributes are correct, as shown in [Figure 5-6](#).

- Check whether the outgoing prefix (**Info**) is added to the bound PSTN user number. The extra-prefix 8 is added, as shown in [Figure 5-6](#).

If no, run the **config modify subscriber unicall dn** <string> **bind pstnnum value** <string> command to change the **Info** value.

- Check whether a validity period (**UseTime**) is set for the binding.

If yes, run the **show timerange** command to view the validity period. You can use the **config modify timerange index** <0-3840> **begintime** <string> **endtime** <string> command to modify the validity period.

- Check whether the interval (**SelectInte**) for sequential ringing is long.

Check whether the product of the value of **selectinterval** and the number of terminals is greater than 60 (seconds). If yes, run the **config modify subscriber unicall dn** <string> **selectinterval** <5-30> command to change the value of **selectinterval**.

**Figure 5-6 ONLY information**

```
[%SoftCo5816(config)]#show subscriber unicall dn 5001

UserSerialNo Dn   SubPBXNo State BusySelectMode PriorityMode RingMode SelectInte
rval
-----
168          5001 0      FAULT all          static    append    12(s)

Terminal bind infomation

Index  Type           Info           Priority UseTime Timeindex Status
-----
0      SIPUE(NATIVE) 5001          0        No      -        FAULT/LOGOUT
1      PSTNDN        85002         0        No      -        -
2      PSTNDN        813788888888 0        Yes     0        -

----- Matched record number : [3] -----

==== Command executed success ! ====
```

- Step 2** Check whether the services enabled for the bound users conflict with the ONLY service. For details about related services, see [5.1.2 ONLY Service](#).
- Step 3** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.
- End

## 5.5 Main Number Service Fault

This topic describes the common main number service and how to troubleshoot the main number service fault.

### 5.5.1 Prompt Message Faults By Main Number Service

This topic describes how to troubleshoot prompt message faults by main number service.

#### Symptom

Common main number service faults are as follows:

- No prompt message is played when the main number is dialed.
- An incorrect prompt message is played when the main number is dialed.

#### Possible Causes

The possible causes are as follows:

- The access code for the VU main number is incorrect.
- The MRU or MRS on the SoftCo9500 is not functioning properly.
- The VU script is imported incorrectly or the VU script is incorrect.
- The VU main number service is incorrectly configured.
- Files in the **voice.zip** package are incorrect.
- The default incoming number is set.

## Procedure

**Step 1** Run the **show prefix dn** *<string>* command to check whether the access code for the VU main number is correct. The value of **If4PSTN** should be **yes**.

If no, run the **config modify prefix dn** *<string>* **if4pstnprefix yes** command to change the access code.

**Step 2** Run the **show board** command on the SoftCo9500 to check whether the MRU or MRS is in **OK** state.

If the status of the MRU or MRS is **FAULT**, proceed as follows to rectify the fault:

1. Remove and then reinstall the MRU or MRS, or replace the MRU or MRS.
2. Log in to the DTU and MRU through the serial port and run the **show version** command to check whether the versions of the DTU and MRU are the same as the version of the MCU. For details on how to log in to the MRU or MRS through the serial port, see How to Connect to the SoftCo Through the Serial Port or by Telnet.

If the version of the MRU or MRS is different from the version of the MCU, upgrade the version.

**Step 3** Check whether the VU script is imported correctly.

In the super user view mode of the SoftCo, run the **debug print cli history info place** *<current|backup>* command to check whether the VU script is imported. You can also rectify the fault by importing the VU script again.

**Step 4** Run the **show prefix callcategory vu** command to check whether the value of **Vug** is the same as the value of **define OperationNO** in the VU script, as shown in [Figure 5-7](#) and [Figure 5-8](#).

By default, the value of **Vug** is **1** for the automatic main number service, and the value of **Vug** is **2** for the conference service. You can set the **Vug** value for any other service to a number ranging from **4** to **59**. For example, the value **8** refers to the password-based call restriction service.

**Figure 5-7** Vug value

```
[%SoftCo5816(config)]#show prefix callcategory vu
Prefix
SubPBXNo Prefix CallAttribute Vug RegGroupId
-----
0        68908888 vuconsole 1 0
==== Command executed success ! ====
```

**Figure 5-8** VU script

```

10  define OperationNO  1
50  define true        1
80  define false      0
100 str groupNo = "0"
200 str transferNum = "8000"
210 int NumLen = 0
220 str calledNum = ""
230 int seatNo = 65535
300 calledNum = ""
301 NumLen = 0
302 PlayCollect 512,512,250,true,32,'#','*','#','#',20,NumLen,calledNum
    
```

**Step 5** Check whether the channel for playing prompt messages in the script is correct, as shown in [Figure 5-8](#).

The value **250** indicates the ID of the channel for playing prompt messages. Use the tool for playing prompt messages to check whether the **250.pcm** file is correct. If the file is incorrect, record the prompt messages again to generate a new **250.pcm**.

**Step 6** Run the **show protocol <at0|cassignal|isup|tup|pra|sip|h323>** command to check whether the default incoming number is set. The default incoming number is set to be 88888888, as shown in [Figure 5-9](#).

If yes, run the **config protocol <at0|cassignal|isup|tup|pra|sip|h323> officeno <0-254>defaultclnum1 0000** command to delete the default incoming number.

**Figure 5-9** Default incoming number

```

[%SoftCo5816(config)]#show protocol sip
SIP
Office IPAddress      DomainName LocalUdpPort LocalTcpPort PeerPort TransportType
-----
0      10.166.50.187 bpx1      5060      5060      5060      UDP
MaxCallNum DefaultCLDNum1 DefaultCLDNum2 CallerPhoneContext CalleePhoneContext
-----
240      88888888      None
DeviceType DTMN      SupportInfo CRBT SupportCliPredeal ImsConnId
-----
OtherDevice refused No      Yes Yes      ---
==== Command executed success ! ====
    
```

**Step 7** See [2.3 Log Analysis](#) to capture the log and rectify the fault.

**Step 8** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.5.2 User Faults By Main Number Service

This topic describes how to troubleshoot user faults by main number service.

## Symptom

A calling party dials a number as prompted, but the call cannot be connected or is connected 10 seconds later.

## Possible Causes

The possible causes are as follows:

- The VU script is imported incorrectly or the VU script is incorrect.
- The digit collection mode on the SoftCo is different from that on the peer device.

## Procedure

**Step 1** Check whether the VU script is imported correctly.

In the super user view mode of the SoftCo, run the **debug print cli history info place** <current|backup> command to check whether the VU script is imported. You can also rectify the fault by importing the VU script again.

**Step 2** Check whether the script is correct, as shown in [Figure 5-10](#).

- Check whether the value of **groupNo** is correct.
- Check whether the value of **transferNum** is correct.
- Check whether the maximum number of digits to be collected (the fifth parameter of the **PlayCollect**) and the number end symbol (the ninth parameter of the **PlayCollect**) is correct.
  - Generally, the value is greater than or equal to the length of the called number. By default, the value is **32**.
  - The default number end symbol is #. You are not advised to set the number end symbol to digits 0 to 9.

For example, if you set the number end symbol to 0 and dials 3000, the SoftCo stops collecting digits when receiving a 0. This leads to call failure.

**Figure 5-10** VU script

```
10  define OperationNO  1
50  define true        1
80  define false      0
100 str groupNo = "0"
200 str transferNum = "8000"
210 int NumLen = 0
220 str calledNum = ""
230 int seatNo = 65535
300 calledNum = ""
301 NumLen = 0
302 PlayCollect 512,512,250,true 32,'#','*','#',#,20,NumLen,calledNum
```

**Step 3** If the SoftCo uses the SIP trunk to connect to the peer device, check whether the SoftCo and the peer device use the same digit collection mode.

Contact the peer engineer to check the digit collection mode and the value of **PayloadTpye**. If the RFC2833 digit collection mode is enabled on the peer device, perform the following operations on the SoftCo:

1. Run the **show cdsp argu** command to view the digit collection mode. The RFC2833 digit collection mode is disabled by default. This means that **RFC2833Flag:0** is displayed.
2. Run the **config cdsp rfc2833switch <on|off>** command in Config mode to enable the RFC2833 digit collection mode.
3. Run the **show cdsp payloadtype** command to check whether the RFC2198 load (the value of **PayloadTpye**) is the same as that of the peer device.  
If the values are different, run the **config cdsp payloadtype rfc2833 value<0-255>** command in Config mode to change the value.

**Step 4** See [2.3 Log Analysis](#) to capture the log and rectify the fault.

**Step 5** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.6 Fax service fault

This topic describes the common main number service and how to troubleshoot the fax service fault.

### 5.6.1 Faults Occurring When Sending or Receiving Faxes

This topic describes how to troubleshoot faults occurring when sending or receiving faxes.

#### Symptom

The faults are as follows:

- No fax can be sent.
- No fax can be received.
- Faxes can be sent successfully only on some fax machines.
- Faxes can sometimes be sent successfully.
- The clock source has not been configured or the clock source quality is low.

#### Possible Causes

The possible causes are as follows:

- The network quality does not meet the requirement.
- Operations on the fax machine are incorrect or the fax machine is faulty.
- The fax machine is incorrectly set.
- Fax modes of the SoftCo and IAD are different.
- The receiving gain and sending gain are incorrectly set.
- Some fax machines are incompatible with gateway devices.

#### Procedure

- Step 1** Verify that the network quality meets the requirement. For details, see [2.5 Network Information Analysis](#).

**Step 2** Verify that operations on the fax machine are correct.

1. Before sending a fax, verify that papers are properly placed to the paper tray, and the fax machine is ready.
2. When sending a fax, press the start button only after hearing the fax confirm tone sent by the other party.
3. Check whether the memory of the fax machine is used up. If the memory is used up, clear the memory.
4. Check whether papers are placed to paper trays of both sender and recipient fax machines. If there are papers on both fax machines, remove the papers from the recipient fax machine.

**Step 3** Verify that the fax machines are correctly set.

1. If no fax can be sent, check whether the pulse dialing mode has been enabled. If this mode has been enabled, change it to the touch-tone dialing mode.
2. If no fax can be received, check whether the function of rejecting faxes has been enabled. If this function has been enabled, disable it.

**Step 4** Check whether the transparent transmission mode has been enabled for the SoftCo, IAD.

- For POTS or SIP users, run the **show subscriber dn**<string>command on the SoftCo to check whether the value of **FaxMode** is set to **BYPASS**. If the value is not **BYPASS**, run the **config modify subscriber dn**<string>**faxmode** <null|bypass|t38> command to change the value.
- For H.248 users, run the **show mg** command on the SoftCo to check whether **FaxMode** is **BYPASS**. If the value is not **BYPASS**, run the **config modify mg domainname**<string>**faxmode** <null|bypass|t38> command to change the value.
- On the IAD, run the **display sip send-capability** command to check whether **pri** is **0** and whether **Mode** is **Transfer**. If **pri** is not **0** and **Mode** is not **Transfer**, run the **sip send-capability fax pri 0 transfer** <G711a|G711u> command to change the values.

**Step 5** Adjust the sending and receiving gains to ensure that faxes can be sent and received successfully.

On the IAD, run the **pstnport attribute set**<string><0-17> command to adjust the gains.

**Step 6** Use the comparison test to locate faults.

1. Change a fax machine, and then verify that faxes can be sent and received successfully.
2. Change a phone line of IAD, and then verify that faxes can be sent and received successfully.
3. Change a trunk line, and then check whether faxes can be sent and received successfully. If the faxes can be sent and received successfully, contact the carrier to locate line faults.
4. If faxes failed to be sent or received when a 32-port IAD is used, replace it with a 4-port IAD and check whether faxes can be sent and received successfully.

A 4-port IAD amplifies signals by default to improve the success rate of sending and receiving faxes.

**Step 7** Use a packet catcher to capture signaling and RTP packets, and then rectify faults. For details on how to use the packet catcher, see How to Use the Wireshark.

1. Check whether a re-invite message exists in the signaling process. If the message does not exist, use the Capsens to convert the captured RTP packets to a voice file, and then analyze the file. If the voice file is empty, analyze the log of the device that sent the

re-invite message to locate the fault. For details on how to use the Capsens, see [8.1 How to use the Capsens](#).

2. Check whether the codec type and packaging duration contained in the re-invite message are the same as those contained in the 200 OK message.

If they are different, change the codec type and packaging duration for the SoftCo fax service. Run the **show system information** command to view the voice codec type, as shown in [Figure 5-11](#).

- To set the voice codec type, run the **config system codetype** `<g711a&g711u&g729&g723>` command.
- To change the priority of the voice codec type, run the **config modify office no**`<0-254>codetype <g711a&g711u&g729&g723>` command.
- To change the packaging duration, run the **config system voip** `<g711a&g711u&g729&g723>ptime <ten|twenty|thirty>` command.

**Figure 5-11** Voice codec type on the SoftCo

```
[%SoftCo9500(config)]#show system information

System Information

      Device Type: SoftCo9500
      DomainName: SS
      Net workmode: Single
      IP Address: 10.166.17.94
      SubMask: 255.255.255.0
      MAC Address: 00-c5-3b-8c-73-0b
      GateWay IP Address: 10.166.17.1
      Host IP Address: 10.166.17.218
      CTIServer IP Address: 10.166.17.195
      BillServer IP Setting: 10.166.17.195:2020
      LogServer IP Address: 10.166.17.222
      Voice Code And Decode Type: G711A/G711U/G729/G723
      PertainBoard Type: 8245 HMU 100M
      Master/Slave Flag: Master
      Running Time: 25 Hours 8 Minutes
      Current Time: Date:2010-05-29 Time:11:51:58
      Time Zone: GMT+08:00

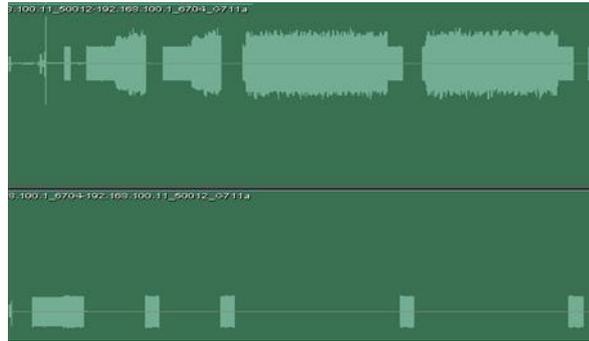
==== Command executed success ! ====
```

3. Check whether the value of **Ptime** in the RTP package that was sent is the same as that in the RTP package that was received. If they are different, contact Huawei technical support.

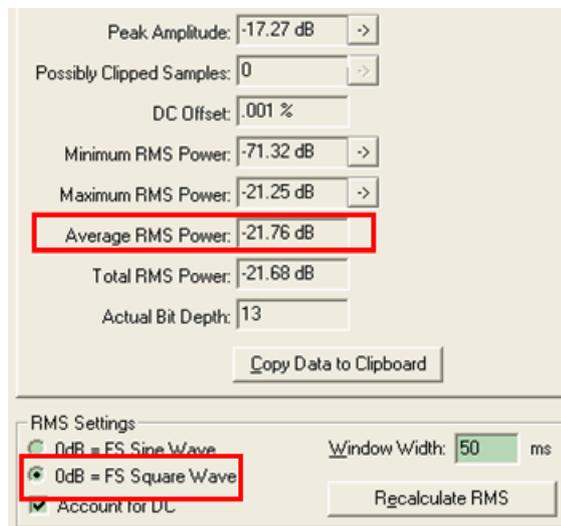
**Step 8** Use the CoolEdit to analyze the fax signals and rectify faults. For details on how to use the CoolEdit, see [8.2 How to use the CoolEdit](#).

- [Figure 5-12](#) compares waves for signals received and sent when an IAD user can send and receive faxes successfully. The average Root Mean Square (RMS) power is about -21 dB, as shown in [Figure 5-13](#).

**Figure 5-12** Waves for fax signals in normal cases



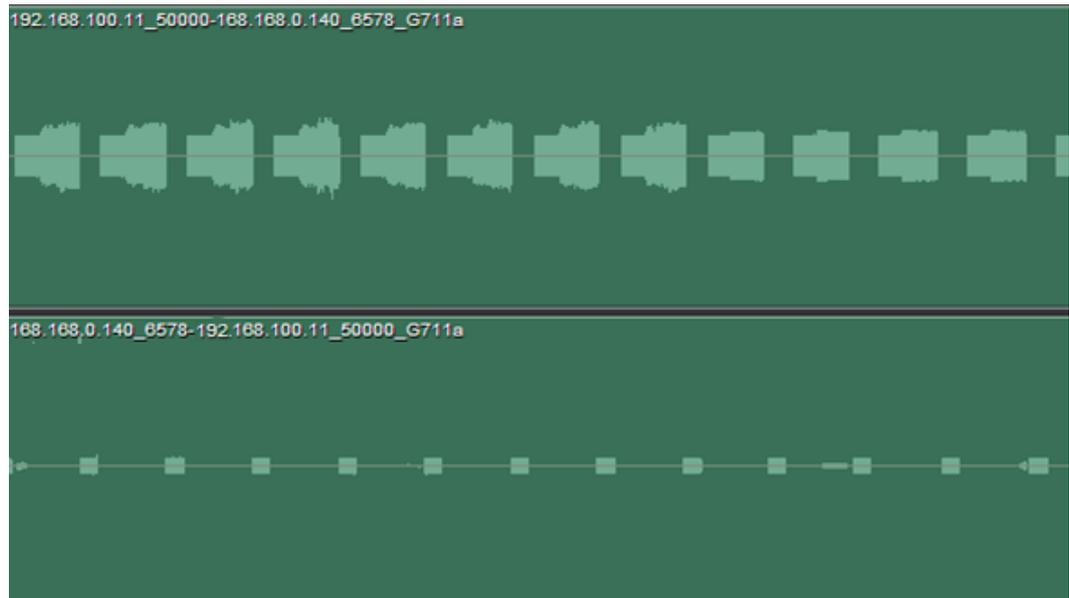
**Figure 5-13** RMS power in normal cases



- **Figure 5-14** compares waves for signals received and sent when an IAD user cannot send faxes.

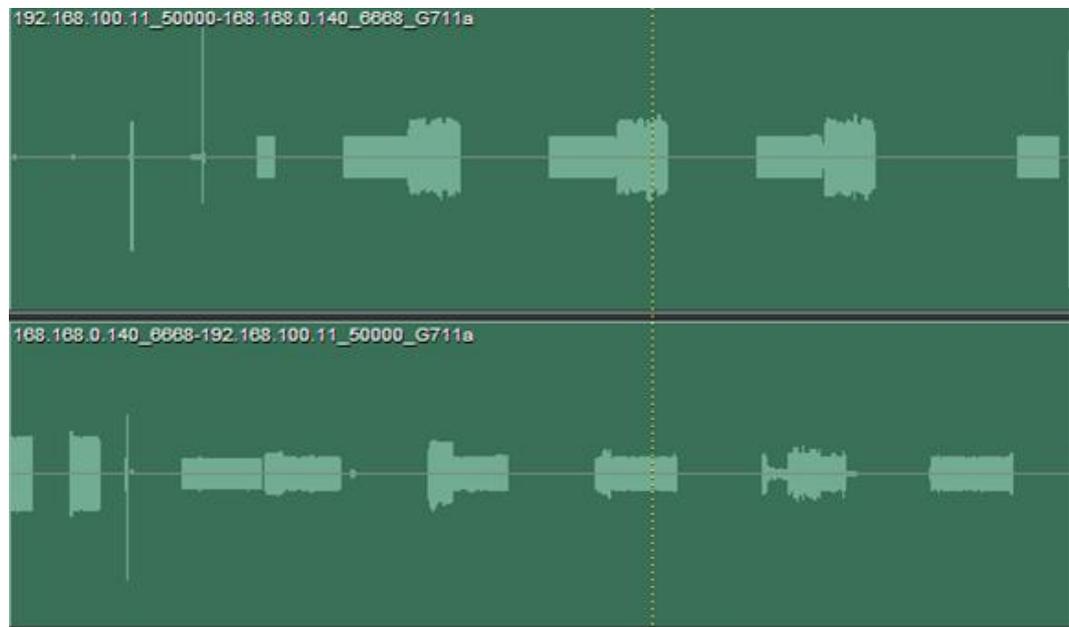
In the chart, **192.168.100.11\_50000-168.168.0.140\_6578\_G711a** is the waves for signals that are sent, and **168.168.0.140\_6578-192.168.100.11\_50000\_G711a** is the waves for signals that are received.

**Figure 5-14** Waves for signals of faxes that cannot be sent



- [Figure 5-15](#) compares waves for signals received and sent when faxes are delayed. This fault is caused by incorrect trunk line connection. If this fault occurs, contact the peer device engineers to check the line.

**Figure 5-15** Waves for signals of delayed faxes



- Step 9** If the Time Division Multiplexing (TDM) trunk is used, see [4.2 E1 Trunk Exception](#) to check whether the clock source has been configured.
- Step 10** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.6.2 Fax Quality Is Low

This topic describes how to troubleshoot the low fax quality.

### Symptom

The symptoms are as follows:

- Images and words are unclear.
- The fax is sent successfully, but the peer party can only receive part of the pages.

### Possible Causes

The possible causes are as follows:

- The network quality does not meet the requirement.
- The SoftCo is not grounded.
- The clock source has not been configured or the clock source quality is low.

### Procedure

- Step 1** Check whether situations such as packet loss, delay, or jitter occur on the IP bearer network. For the detailed procedure, see [2.5 Network Information Analysis](#).
- Step 2** Check whether the SoftCo is grounded.
- Step 3** If the Time Division Multiplexing (TDM) trunk is used, see [4.2 E1 Trunk Exception](#) to check whether the clock source has been configured.
- Step 4** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.

----End

## 5.7 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

### 5.7.1 The System Asks the User to Press 0 to Forward the Call to the Reception Desk or Press 5 to Enter the VoIP Process When a User Dials the Main Number

This topic describes how to troubleshoot the system asks the user to press 0 to forward the call to the reception desk or press 5 to enter the VoIP process.

### Symptom

When the peer user dials the main number, the system asks the user to press 0 to forward the call to the reception desk or press 5 to enter the VoIP process. The peer user hears the voice prompt "Your call has been restricted."

## Possible Causes

The VU script is configured correctly. Run the **show trunkgroup no <0-6120>** command. You can find that the value of **BasicOutgoing** is **INTER**.

## Procedure

- Run the **config modify trunkgroup no 31 outgoingright local** command to add local call right.

----End

## 5.7.2 The Called Phone Rings Abnormally After the Calling Party Hangs up the Phone

This topic describes how to troubleshoot the fault that the called phone rings abnormally after the calling party hangs up the phone.

### Symptom

An IMS user or an intra-office user dials the switchboard number to call another intra-office user. The called intra-office user does not answer. After the called user hangs up the phone, the called user's phone rings.

### Possible Causes

The IMS domain name set on the SIP trunk is different from that set in the IMS connection parameters.

### Procedure

- Step 1** Run the **show peercomponent** command to check the domain name set on the SIP trunk (the value of **DomainName**).
- Step 2** Run the **show ims connectpara** command to check the domain name set in the IMS connection parameters (the value of **DomainName**).
- Step 3** In Config mode, run the **config modify peercomponent domainname<string>ip <A.B.C.D>tknum<1-1200>** or **config modify ims connectpara domainname<string>ip <A.B.C.D>officeselectcode<0-254>mode <master|slave>** command to set domain names to the same.

----End

## 5.7.3 One or More User Phones Fail to Ring When the ONLY Service Is Used

This topic describes how to troubleshoot an ONLY service fault.

### Symptom

After a user dials the ONLY number, one or more user phones fail to ring if the sequential ringing service is enabled.

## Possible Causes

1. Run the **show subscriber dn <string>type all** command. If the value of **PARALLEL\_RING\_SERVICE** is **Y**, the sequential ringing service is enabled.
2. Run the **show ringservice dn <string>** command to view the numbers used in the sequential ringing service and run the **show subscriber unicast dn <string>** command to view the numbers in the ONLY service. You can find that the same number is used in the two services. The bound user phones do not ring because the sequential ringing service conflicts with the ONLY service.

## Procedure

- Run the **config delete ringservice dn <string>** command to disable the sequential ringing service.

----End

## 5.7.4 Call Transfer Service of a SoftCo User Is Abnormal When the SoftCo Is Connected to the IMS

This topic describes how to troubleshoot a fault that occurs when the SoftCo is connected to the IMS.

### Symptom

After a SoftCo user presses the hookflash to transfer a call to another SoftCo user, the two SoftCo users cannot hear each other.

### Possible Causes

The SoftCo is connected to the IMS through the SBC and all services are provided by the IMS. The SBC does not support media stream, so the calling party and the called party cannot hear each other.

### Procedure

- Upgrade the SoftCo version to V100R002C04. For details, see the *SoftCo Upgrade Guide*.

----End

## 5.7.5 No Prompt Message is Played When the Mobile User Rejects the Call

This topic describes how to troubleshoot the One Number Link You (ONLY) service faults.

### Symptom

When the called party is a mobile user and rejects a sequential ringing service connection, no prompt message is played.

## Possible Causes

A possible cause is that the mobile user is not the last user for the sequential ringing service.

## Procedure

- Run the **show subscriber unicall dn** *<string>* command to check whether the value of **Priority** in **Terminal bind information** is **3**.

If no, run the **config modify subscriber unicall dn** *<string>***modbind pstnnum value** *<string>* **priority 3** command to change the mobile user to the last user.

----End

## 5.7.6 IMS User Cannot Be Called in the Simultaneous Ringing Service (ONLY)

This topic describes how to troubleshoot a simultaneous ringing service fault.

## Symptom

SoftCo ONLY number B is bound to IMS number C. When IMS user A dials ONLY number B, the phone of IMS number C does not ring. However, SoftCo ONLY number B can call IMS user C. In addition, when other intra-office call B, the phone of IMS number C rings.

## Possible Causes

The multiple call function is enabled for the IMS account bound to the ONLY number.

## Procedure

- Contact the peer office's engineer to enable the multiple call function for the IMS account on the IMS.

----End

## 5.7.7 Fault Occurs When an IMS User Is Connected in the Simultaneous Ringing Service (ONLY)

This topic describes how to troubleshoot a fault that occurs when an IMS user is connected in the simultaneous ringing service.

## Symptom

SoftCo ONLY number B is bound to IMS number C. When IMS user A dials ONLY number B, the phone of IMS number C rings. When picking up the phone, IMS user C hears the busy tone or the call restriction prompt message.

## Possible Causes

If the session border controller (SBC) proxy is used, the calling party and called party cannot hear each other.

## Procedure

- Contact the peer office's engineer to enable the functions of multiple call, transfer upon inquiry, and call holding for the corresponding IMS account.

----End

## 5.7.8 IMS User Cannot Transfer Calls

This topic describes how to troubleshoot a fault that occurs when an IMS user attempts to transfer calls.

### Symptom

When an IMS user dials the VU number and then enters another IMS user number as prompted, call transfer fails.

### Possible Causes

- The IMS registration group ID is not added to the VU prefix.
- The user to whom the call is to be transferred does not exist or the user is busy.
- The call transfer service is disabled on the IMS.

## Procedure

**Step 1** Run the **show prefix dn <string>** command to check whether the registration group ID (**RegGroupId**) is added to the VU prefix.

If no, add the registration group ID to the VU prefix.

**Step 2** On the IMS, check whether the user to whom the call is to be transferred exists or whether the user is idle.

**Step 3** On the IMS, check whether the CFB service and call holding service is enable.

If no, contact the IMS's engineer to enable the CFB service and call holding service for the registration group ID corresponding to the VU.

**Step 4** If the fault persists, contact Huawei customer service center or the local office.

----End

## 5.7.9 Fax Fails When the SoftCo Is Connected to the IMS

This topic describes how to analyze and troubleshoot the fax fault when the SoftCo is connected to the IMS.

### Symptom

Faxes can be sent or receive only.

### Possible Causes

The media attribute does not match the SoftCo so that the fax is abnormal.

## Procedure

- Step 1** Use the packet capture tool to capture the INVITE SDP message from the peer device, then discover the SDP message's media attribute is **modem**, as shown in [Figure 5-16](#).

**Figure 5-16** SDP message

```

    Message body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): HuaweiSoftCov100R002 12564 12566 IN IP4 10.120.1.146
      Session Name (s): sip Call
      Connection Information (c): IN IP4 10.120.1.146
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 6980 RTP/AVP 8 0
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a):ptime:20
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a):ptime:20
      Media Attribute (a):silenceSupp:off - - -
      Media Attribute (a): modem
    
```

- Step 2** Run the **show softargu** command to view the software parameter ID of **fax and modem sdp attr mode** (the value of **Tpye**).

```

[%SoftCo5816(config)]#show softargu

All Arguments in SoftArgu Table

Type Value      Meaning
-----
0  --          System max user number
1  --          System max cpu ration
...
255 --          fax and modem sdp attr mode [0: FAX/MODEM, 1: X-FAX/X-MODEM]
    
```

- Step 3** Run the **config softargu type 255 value 0** command to modify the media attribute to **FAX/MODEM**.

----End

## 5.7.10 Faxes Cannot Be Sent Using T.38

This topic describes how to troubleshoot a fault that occurs when faxes are sent using T.38.

### Symptom

The MRU and MRS are configured on the SoftCo. User A sends a fax to user B. The fax fails to be sent using T.38.

### Possible Causes

The default values of **conference** parameter for the MRS and MRU are both **yes**. If both the MRS and MRU are configured on the SoftCo, the MRS is used when a user sends a fax. The MRS does not support T.38 and the fax fails to be sent.

## Procedure

- Run the **config modify board slot <2-9> type mru conference no** command to change the value of **conference** parameter for the MRU to **no**.

----End

---

# 6 System Device Faults

---

## About This Chapter

System device faults usually occur in the power supply system or during the upgrade.

### 6.1 Faults Occurring During the Upgrade

This topic describes how to troubleshoot a fault occurring during the upgrade.

### 6.2 Power Supply Exception

This topic describes how to troubleshoot power supply exceptions.

### 6.3 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

## 6.1 Faults Occurring During the Upgrade

This topic describes how to troubleshoot a fault occurring during the upgrade.

### Symptom

The following faults frequently occur during the upgrade:

- Data cannot be backed up.
- The version software cannot be loaded.

### Possible Causes

Junk data exists in the Flash memory or the Flash memory is insufficient.

### Procedure

- Step 1** Connect to the SoftCo through the serial port. See How to Connect to the SoftCo Through the Serial Port or by Telnet.
- Step 2** Start the BOOT1 mode and run the **KDF** command to delete all the data from the Flash memory, as shown in [Figure 6-1](#).

**Figure 6-1** Deleting data

```
[BOOT1]: KDF
[BOOT1]: Are you sure to erase all data on flash? [Y/N]y
Total blocks : 0x100. The current block 0x8 erased OK.
```

**Step 3** Run the **boot** command to restart the SoftCo.

**Step 4** Upgrade the SoftCo again. For details, see the *SoftCo Upgrade Guide*.

----End

## 6.2 Power Supply Exception

This topic describes how to troubleshoot power supply exceptions.

### Symptom

The SoftCo9500 is powered on. After you press the Mute button, the buzzer keeps ringing.

### Possible Causes

The possible causes are as follows:

- The power supply is abnormal.
- The power module is incorrectly connected to the power cable.
- The external power supply is abnormal.
- The power module or the filler panel is faulty.

### Procedure

**Step 1** Check whether the RUN indicator is normal. Take measures according to the indicator status, which is described in [2.2 Alarm Analysis](#).



#### NOTE

If the power indicator blinks abnormally, check whether the voltage is low.

**Step 2** Check whether the power module and power cable are fastened. Remove and reconnect the power cable and power on the device to check whether the alarm is cleared.

- If yes, check whether alarms are still displayed.
  - If no alarms are generated but the power supply remains abnormal, the line voltage may be instable or the short-term self-protection function is enabled.
  - If an alarm is generated, output over-voltage has occurred.
- If no, replace the power module.

If the alarm still exists, the external power supply may be faulty or the power supply alarm is reported mistakenly.

- Step 3** Check whether power modules and filler panels in other slots are fastened and whether all idle power modules are powered on. If no, fasten power modules and filler panels and power on idle power modules.
- Step 4** Fasten and power on the power modules. Then check whether an alarm is generated. If so, replace the power modules or filler panels.
- Step 5** If the fault persists, see [1.6 Technical Support](#) to obtain technical support.
- End

## 6.3 Typical Cases

This topic describes the fault symptom, cause analysis, and fault troubleshooting.

### 6.3.1 Voice Service Fails Frequently After the SoftCo Is Upgraded

This topic describes how to troubleshoot a fault that occurs after the SoftCo is upgraded.

#### Symptom

After the SoftCo is upgraded, users frequently report failures such as call drop, registration failure, and low voice quality.

#### Possible Causes

The network is congested due to inconsistent working modes of the SoftCo's service network ports and the switch.

#### Procedure

- Step 1** View the alarm information through the OMU. You may find that alarms such as **SIP terminal failure** are reported continuously even at night when there is no or low traffic. In addition, the phone sets do not receive a response to the re-registration request and network packets are lost.
- Step 2** View the status of the Ethernet switch that connects to the SoftCo. You can find that the Ethernet switch works in half-duplex mode at 100 Mbit/s, whereas the SoftCo works in full-duplex mode at 100 Mbit/s. The packet loss rate is high due to the different working modes of the SoftCo and the Ethernet switch.
- Step 3** Change the working mode of the Ethernet switch because the network adapter of the SoftCo does not support automatic negotiation. For example, run the following command to change the working mode of the switch in a specific office:

```
lsw set portno 17 status enable auto on speed speed100 duplex full flowctrl disable
```

#### NOTE

- The MCU has two network ports. The preceding fault occurs only on network port 0. It is recommended that you use network port 1. If network port 0 must be used, perform step 3 to rectify the fault.
- To configure the command to change the working mode of a switch, refer to the documents that are delivered with the actual switch model.

----End

## 6.3.2 Uploading or Downloading a File Fails

This topic describes how to troubleshoot a fault that occurs when a user uploads or downloads a file.

### Symptom

The system displays an error message when a user uploads a file to the TFTP server or downloads a file (for example, a license file) from the TFTP server.

### Possible Causes

- The **license.dat** file on the TFTP server is named incorrectly.
- The value of **hostip** on the SoftCo is different from the IP address of the TFTP server.
- If downloading a file fails, the file's directory on the TFTP server may be incorrect or the file to be downloaded may be unavailable.

### Procedure

**Step 1** Check whether the **license.dat** file on the TFTP server is named correctly. The correct file name is **license.dat**.

**Step 2** Start the Config mode of the SoftCo and run the **config system hostip <A.B.C.D>** command to set a correct IP address for the TFTP server.

**Step 3** Check whether the loading directory on the TFTP server is correct.

----End

## 6.3.3 Pinging Another Device from the SoftCo Fails

This topic describes how to troubleshoot a fault that occurs when another device is pinged from the SoftCo.

### Symptom

Another device cannot be pinged from the SoftCo. Assume that the device is A.

### Possible Causes

- The network cable is disconnected or faulty.
- The IP address or subnet mask of the SoftCo is incorrect.
- After the IP address of the SoftCo is changed, the system is not restarted. As a result, the changed IP address does not take effect.

### Procedure

**Step 1** Verify that device A and the switch are connected and the network cable is normal.

**Step 2** Verify that the SoftCo and the switch are connected and the network cable is normal.

**Step 3** Run the **show system information** command to check the network configuration of the SoftCo.

- If the IP address of the SoftCo and that of device A are on different network segments, change the IP address of the SoftCo to the same network segment as device A.
- If the IP address of the SoftCo and that of device A are on the same network segment, the IP address of the SoftCo does not take effect after the IP address is changed. In this case, restart the system to make the IP address take effect.

----End

# 7 Terminal and gateway faults

---

## About This Chapter

Terminal and gateway device include IAD and IP Phone.

### 7.1 IP Phone

This topic describes the common methods for locating and rectifying IP Phone faults.

### 7.2 IAD

This topic describes the common methods for locating and rectifying IAD faults.

## 7.1 IP Phone

This topic describes the common methods for locating and rectifying IP Phone faults.

### 7.1.1 IP Phone Cannot Be Started

This topic describes how to troubleshoot a starting fault of IP Phone.

#### Symptom

An IP phone cannot be started and nothing is displayed on the screen.

#### Possible Causes

The software of the IP phone is faulty.

#### Procedure

- To recover the software on the ET525, do as follows:
  1. Connect the PC and IP phone to the same hub. Check and ensure that only one network adapter is installed on the PC. If more than one network adapter is installed, enable one network adapter and disable the others.
  2. Set the IP address of the PC to **168.75.215.188**, the subnet mask to **255.255.0.0**, and the IP address of the gateway to **168.75.215.188**.

3. Start the Trivial File Transfer Protocol (TFTP) server on the PC, set the load directory, and start the TFTP service. Then place the upgrade software to the load directory.
4. Press \*+# on the IP phone, cut off the power supply, and restore the power to start the IP phone. The IP phone will download software from the TFTP server.

----End

## 7.1.2 IP Phone Is Disconnected from the Network

This topic describes how to troubleshoot disconnect fault of IP Phone.

### Symptom

The network status indicator light or screen display is not normal, for example, the T-shaped icon in the upper-left corner of the screen is blinking, or the screen displays messages such as "Check Network Connection", "Network Starting", or "Configuration".

### Possible Causes

- The LAN interface is disconnected from the switch.
- Network parameters such as IP addresses are set incorrectly.

### Procedure

**Step 1** Check and ensure that the LAN interface is connected to the switch.

**Step 2** Check and ensure that fields such as IP addresses are set correctly. For detailed operation, see the related IP Phone document.

----End

## 7.1.3 Web Pages Cannot Be Configured

This topic describes how to troubleshoot configuring fault of IP Phone.

### Symptom

After logging in to the Web page, you cannot configure pages including the SIP account page and the advanced parameters page.

### Possible Causes

When you log in to the Web page as an ordinary user with the password **123**, you can only view statuses and change network parameters. You, however, cannot set advanced parameters such as the SIP account.

### Procedure

- Log out of the Web page, and then log in to the Web page again as an administrator with the password **admin**.

----End

## 7.1.4 IP Phone Can Receive Incoming Calls but Cannot Make Outgoing Calls

This topic describes how to troubleshoot none but receiving incoming calls fault of IP Phone.

### Symptom

On the ET522 or ET322, you can receive incoming calls but cannot make outgoing calls.

### Possible Causes

The fault occurs when the proxy server is not set on the ET522 or ET322. These two types of IP phones can work normally even if the registration server is not set. The proxy server, however, must be set so that they can work normally.

### Procedure

- Step 1** Log in to the Web page of the IP phone with an administrator account, and then access the SIP configuration page.
  - Step 2** Set the SoftCo as the proxy server. For the ET522, enter the IP address of the SoftCo in the **SIP Proxy Server** text box.
- End

## 7.1.5 IP Phone Can Receive Outgoing Calls but Cannot Make Incoming Calls

This topic describes how to troubleshoot none but making outgoing calls fault of IP Phone.

### Symptom

You can make outgoing calls but cannot receive incoming calls on the ET523.

### Possible Causes

When the do-not-disturb (DND) function is enabled on the ET523, the incoming calls are rejected.

### Procedure

- Check the DND icon. If the icon blinks, you can infer that the DND function is enabled. When the ET523 is not used, press **MUTE/DEL** to disable the DND function.
- End

## 7.1.6 IP Phone Cannot Provide Two-Stage Dialing

This topic describes how to troubleshoot the two-stage dialing fault of IP Phone.

## Symptom

An IP phone cannot provide the two-stage dialing.

## Possible Causes

The SoftCo supports two dual-tone multifrequency (DTMF) signal collection modes (that is, in-audio and Info); therefore, the DTMF signal sending modes of the IP phone must be set to in-audio and Info modes.

## Procedure

- Check DTMF signal sending modes of the IP phone and ensure that in-audio and Info modes are selected. For the ET523, access the SIP configuration page (that is, the **ACCOUNT** page), and then select **in-audio** and **via SIP INFO** in **Send DTMF**.



### NOTE

When the IP phone supports G711A codec, it is recommended that you set top priority for G711A, that is, set **choice 1** in the **Preferred Vocoder** area to **G.711A**.

----End

## 7.1.7 IP Phone Cannot Obtain Time from the NTP Server

This topic describes how to troubleshoot obtaining time fault of IP Phone.

## Symptom

When a PC is used as the NTP server, an IP phone cannot obtain time from the PC.

## Possible Causes

- The fault occurs on the ET523 of V1.1.2.4.
- The firewall is installed on the PC, so NTP packets sent by the IP phone are intercepted.

## Procedure

**Step 1** Check the version of the ET523 on the **STATUS** page. If the version is V1.1.2.4, upgrade it to a later version.

**Step 2** If the firewall is installed on the PC, rectify the fault in one of the following ways:

- Disable the firewall on the PC.
- On the firewall, configure rules so that NTP packets can pass. Add an exception with the port number 123 (a port number frequently used by the NTP server) and the protocol UDP.

----End

## 7.2 IAD

This topic describes the common methods for locating and rectifying IAD faults.

## 7.2.1 The System Time Restores to the Factory Setting After the IAD Is Restarted

This topic describes how to troubleshoot the factory setting fault of IP Phone.

### Symptom

After the IAD is restarted, the time displayed on the phone set connected to the IAD changes to the factory setting time of the IAD.

### Possible Causes

The IAD does not use the SNTP function to synchronize the time with the SoftCo.

### Procedure

**Step 1** Start the SNTP service of the SoftCo.

```
[%SoftCo9500(config)]#start sntpserver
```

**Step 2** Start the SNTP client on the IAD to keep time synchronization with the SoftCo.

```
TERMINAL(config)#sntp server address 192.168.10.10 //Sets the IP address of the SNTP server. In this example, the SoftCo is used as the SNTP server. The IP address is 192.168.10.10.
```

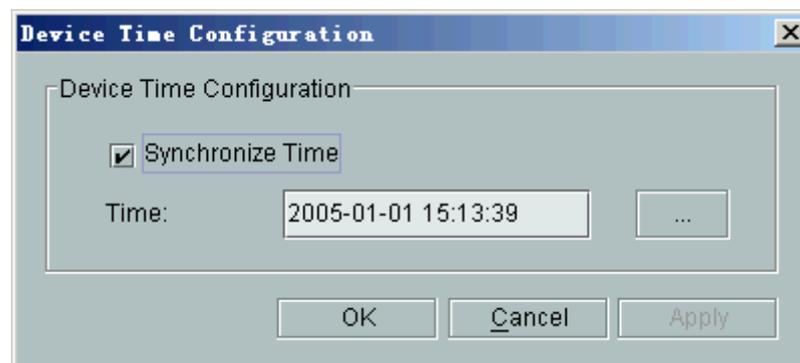
```
TERMINAL(config)#sntp time-zone + 8 //Sets the time zone to UTC+8. You need to change the value according to the actual situation.
```

```
TERMINAL(config)#sntp interval 60 //Sets the time interval for synchronization. In this example, the value is 60s.
```

**Step 3** If many IADs are used, and you have the UCEMS tool, you can enable the SNTP function of the IADs in batches on the UCEMS.

Right-click the IAD to be configured in a specific area and select **Advanced Parameters > Device Time Configuration** from the shortcut menu. In the dialog box that is displayed, select **Synchronize Time**, as shown in [Figure 7-1](#).

**Figure 7-1** Enabling the SNTP function of the IADs in batches through the UCEMS



----End

## 7.2.2 Call Is Abnormal If an IAD User Does Not Dial #

This topic describes how to troubleshoot a fault that occurs when an IAD user does not dial the pound key (#).

### Symptom

An IAD user does not dial the pound key (#) after dialing a user number, resulting in an excessively long connection time.

### Possible Causes

The IAD does not support the function of reporting a number digit by digit.

### Procedure

**Step 1** Run the **show version** command check whether the SoftCo support the function of reporting a number digit by digit. Versions later than V100R002C02B089 support the function of reporting a number digit by digit.

**Step 2** Set a digitmap so that the user does not need to dial the pound key (#).

1. Run the **show softargu** command on the SoftCo to view the value of **Type** corresponding to **sip support iad no pond switch**.

```
[%SoftCo5816(config)]#show softargu
```

```
All Arguments in SoftArgu Table
```

```
Type Value      Meaning
```

```
----
```

```
-----
```

```
0  --          System max user number
```

```
1  --          System max cpu ration
```

```
...
```

```
282 --          sip support iad no pond switch[1:(open) 0:close default:0]
```

Run the **config softargu type 282 value 1** command to open the parameter.

2. Depending on the SoftCo version, you can use either of the following methods to set a digitmap:

- If the SoftCo version is V100R002C02B089 or later, run the **config add digitmap protocol sip value <string> sendmapswitch on** command on the SoftCo.

Set **sendmapswitch** to **on**, indicating that the SoftCo automatically sends the digitmap to the IAD.

- If the SoftCo version is earlier than V100R002C02B089, run the **sip digitmap<string>** command on the IAD.

----End

## 7.2.3 Dialing Error Message Is Played When an IAD User Dials an Extension Number

This topic describes how to troubleshoot a fault that occurs when an IAD user dials an extension number.

## Symptom

An IAD user makes a call to the PSTN switchboard, and then dials an extension number when prompted. A voice message indicating that the dialed number is incorrect is played.

## Possible Causes

The RFC2833 dialing mode is not enabled for the IAD and SoftCo.

## Procedure

- Step 1** Run the **sip send-capability 2833** command on the IAD to enable the RFC2833 dialing mode.
- Step 2** Run the **config cdsp rfc2833 switch on** command on the SoftCo to enable the RFC2833 dialing mode.
- End

## 7.2.4 IAD User Hears Beeps During a Call

This topic describes how to troubleshoot a fault that occurs during a call.

### Symptom

An IAD user hears beeps during a call.

### Possible Causes

The RFC2833 digit collection mode is enabled for the IAD.

### Procedure

- If the digit collection function is not affected, run the **undo sip send-capability 2833** command on the IAD to disable the RFC2833 digit collection mode.
- End

## 7.2.5 Low Volume for an IAD User

This topic describes how to troubleshoot a fault that causes low voice volume.

### Symptom

The voice volume for an IAD user is low.

### Possible Causes

The phone volume is low.

### Procedure

- Step 1** Run the **dsp send-gain<2,64>** command to adjust the DSP transmit gain.

In the command, <2,64> indicates the gain ranges from -31 dB to 0 dB, 0.5 dB is to be added for each level.

**Step 2** Run the **dsp recv-gain<2,64>** command to adjust the DSP receive gain.

In the command, <2,64> indicates the gain ranges from -31 dB to 0 dB, 0.5 dB is to be added for each level.

----End

## 7.2.6 Call Ends When the IAD user Is Picked Up in Hands-free Mode

This topic describes how to troubleshoot a fault that occurs when an IAD user picks up the phone in hands-free mode.

### Symptom

An IAD user picks up the phone after hearing that the peer user has picked up the phone in hands-free mode. When the IAD user hears the hookflash prompt tone, the call immediately ends.

### Possible Causes

Due to the user's phone problem, the IAD mistakenly assumes that the call has ended and reports a hookflash event.

### Procedure

**Step 1** Run the **devasi parameter set ?** command to view the maximum and minimum hookflash durations for the IAD, as shown in [Figure 7-2](#).

**Figure 7-2** Maximum and minimum hookflash durations

```
TERMINAL (advanced-config)#devasi parameter set ?
-----
Advanced mode command
-----
{<U>}<0, 21>      Parameter ID No.
                  0 : Feedback Voltage
                  1 : Interface Impedance
                  2 : A/u Law
                  3 : Hooking up limit (*10ms) Default 30
                  4 : Hooking down limit (*10ms) Default 8
                  5 : 12KC/16KC Signal
                  6 : 12KC/16KC Signal Power
```

**Step 2** Run the **devasi parameter set 3 80** command to increase the maximum hookflash duration.

**Step 3** Run the **devasi parameter set 3 50** command to decrease the minimum hookflash duration.

----End

## 7.2.7 Echoes Can Be Heard When an IAD User Calls a PSTN User

This topic describes how to troubleshoot a fault that occurs when an IAD user calls a PSTN user.

### Symptom

Echoes can be heard when an IAD user calls a PSTN user.

### Possible Causes

- The echoes are acoustic feedback from the user's microphone.
- The EC function is disabled or the mute detection function is enabled on the IAD.
- The receive gain or transmit gain on the IAD is set incorrectly.
- The EC parameter of the peer gateway is set incorrectly.

### Procedure

**Step 1** Check whether the echoes are acoustic feedback from the user's microphone.

The PSTN user covers the microphone to check whether the echoes disappear. If yes, the echoes are acoustic. Acoustic echoes generally occur when the hands-free function is enabled. To prevent acoustic echoes, disable the hands-free function.

**Step 2** Check whether the EC is set correctly.

Run the **show system voip argu algo g711** command on the SoftCo. If the EC on the IAD is set incorrectly, modify the setting as required.

- If the PSTN user hears echoes, run the **dsp ec-enable on** command on the IAD to enable the EC function and run the **dsp vad-enable off** command to disable the mute detection function.
- If the IAD user hears echoes, run the **show system voip argu algo g711** command on the SoftCo to check whether the value of **EchoCancel** is **ON**. If no, Run the **config system voip algo g711 echocancel on** command to enable the EC function on the SoftCo.

**Step 3** Run the **pstnport attribute set** command on the IAD to reduce the receive gain and transmit gain.

- If the transmit gains of IAD108H and IAD132H are both 0 dB, adjust the gains to -3 dB.
- If the receive gain of IAD108H is -3.5 dB, adjust the gain to -7 dB to -8.5 dB.
- If the receive gain of IAD132E(T) is 0 dB, adjust the gain to -3.5 dB to -7 dB.



#### NOTE

Do not reduce the receive gain or transmit gain too much, or the voice may be intermittent or excessively low. The default gains are recommended.

**Step 4** If the fault persists after you adjust the port gains, on the IAD, run the **dsp recv-gain** command to adjust the DSP receive gain, and run the **dsp send-gain** command to adjust the DSP transmit gain.

**Step 5** Contact the peer office's engineers to check whether the EC parameter of the peer gateway is set correctly.

If no, ask the peer office's engineers to deal with fault.

----End

# 8 Common Tool

## About This Chapter

You can make use of common tool to troubleshoot the fault.

### 8.1 How to use the Capsens

You can download Capsens tool from internet.

### 8.2 How to use the CoolEdit

You can download CoolEdit tool from internet.

## 8.1 How to use the Capsens

You can download Capsens tool from internet.

### Context

Use Capsens to change RTP packet to wav file.



#### NOTE

The detail direction of Capsens tool for use see Capsens-related document.

### Procedure



**Step 1** Double-click **Capsens.exe**, run Capsens.

**Step 2** Choose **Tool > CAP2WAV**, the **Cap file convert to wav file** is displayed.

**Step 3** Click , choose the relevant file which need to convert.

**Step 4** Click **Convert** to change .cap to .wav.

----End

## 8.2 How to use the CoolEdit

You can download CoolEdit tool from internet.

### Context

The CoolEdit can be used to the frequency analysis of fax sending and receiving.



#### NOTE

The detail direction of CoolEdit tool for use see CoolEdit-related document.

### Procedure



**Step 1** Double-click , run CoolEdit.

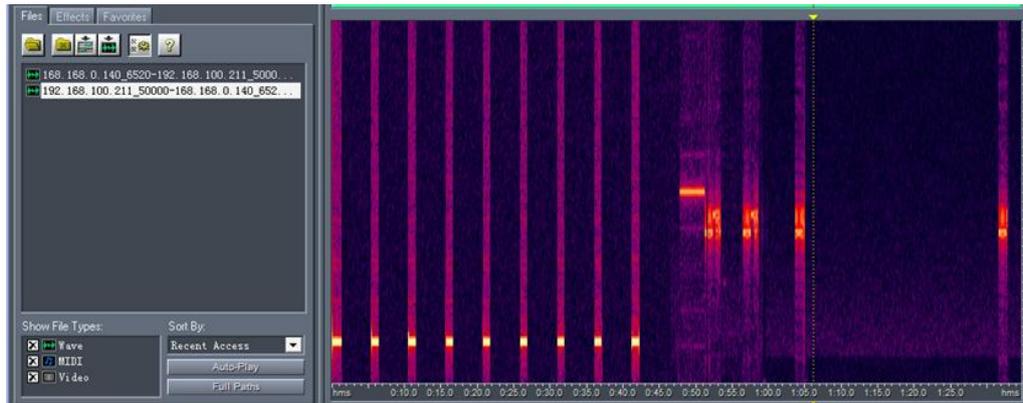
**Step 2** In the main menu, choose **File** > **Open**. The **open** dialog is displayed. Then choose **.wav** file. the waveform is displayed in the main interface, as shown in [Figure 8-1](#).

**Figure 8-1** Wave spectrum



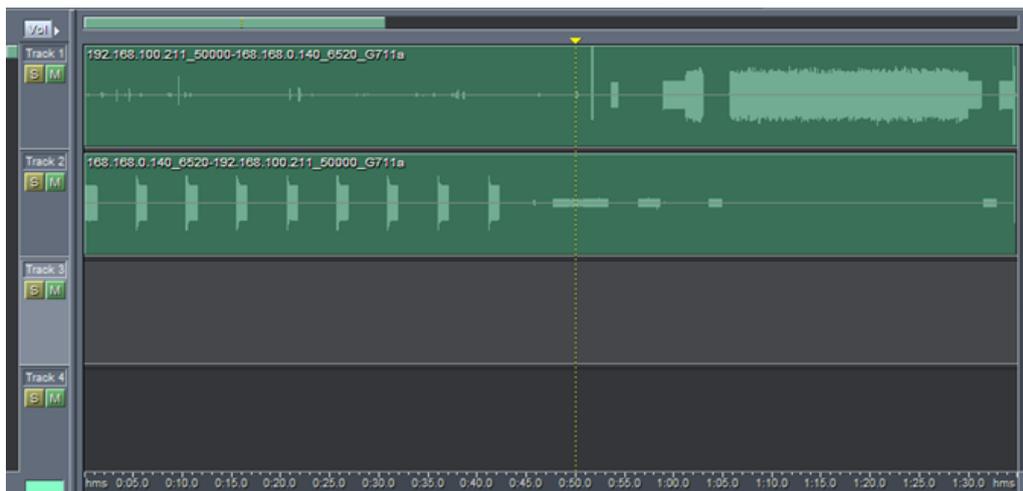
**Step 3** In the toolbar click , the frequency spectrum is displayed, as shown in [Figure 8-2](#).

**Figure 8-2** Frequency spectrum



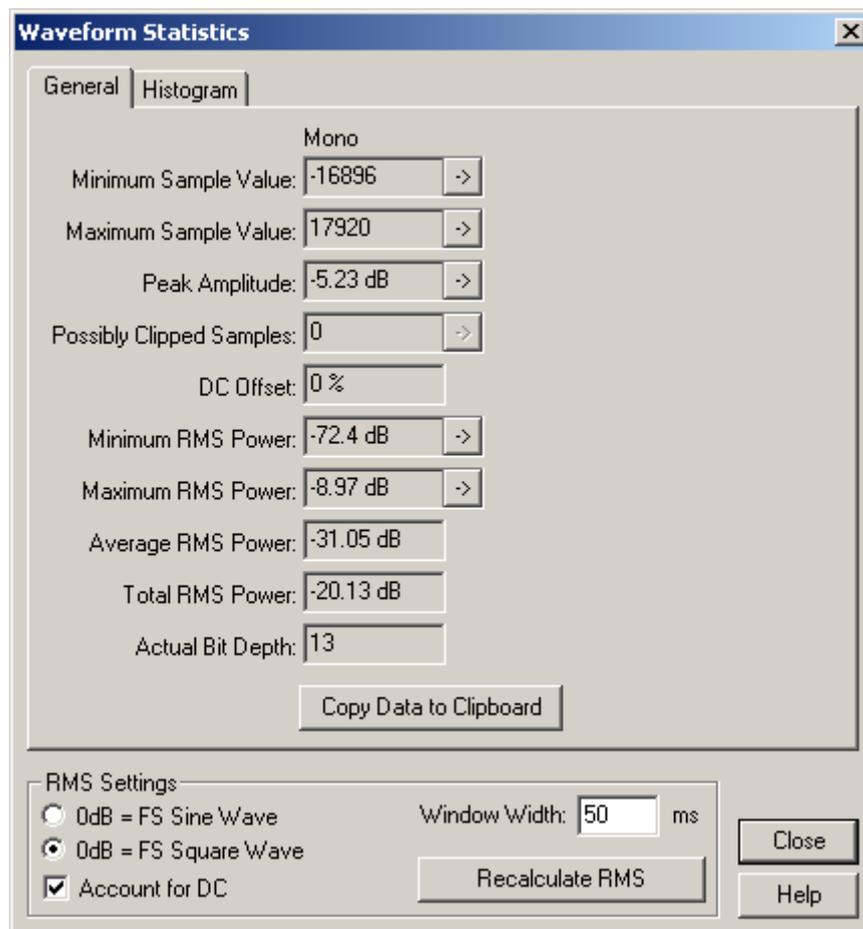
**Step 4** In the toolbar click  to contrasting send and receive signal, as shown in [Figure 8-3](#).

**Figure 8-3** Contrasted signal spectrum



**Step 5** In the main menu, choose **Analyze > Statistics**, the **Waveform statistics** is displayed, as shown in [Figure 8-4](#).

**Figure 8-4** Waveform statistics



----End