Voice Feature Guide

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Preface

Purpose

This document describes voice feature principles and configuration and maintenance guide for Huawei access products where voice can be applied, providing a reference for network design, network entry tests, and network maintenance.

Involves Products and Versions

- This document does not provide feature specifications for specified product versions. If such information is required, go to the **Feature Specifications Query Tool** to see detailed feature specifications and limitations.
  - Carrier
  - Enterprise
- This document uses the MA5600T/MA5603T/MA5608T V800R019C10 as an example to describe these differences. For the products and versions supporting voice, voice principles, configuration logic, and maintenance and diagnosis methods are basically the same. The only difference lies in sub-features and configuration commands/parameters. For details about a specified product version, see the *Product Documentation* of the desired version.
- AGs and MGs mentioned in the document are Huawei access devices.

  The following table lists the products where this document can be applied to.

Product	Version
All products supporting voice, such as MA5600T/MA5603T/MA5608T, MA5800C, MA5616, MA5818, and MA5612	All versions

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description

Symbol	Description
⚠ DANGER	Indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.
⚠ WARNING	Indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.
⚠ CAUTION	Indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury.
NOTICE	Indicates a potentially hazardous situation which, if not avoided, could result in equipment damage, data loss, performance deterioration, or unanticipated results. NOTICE is used to address practices not related to personal injury.
🕮 NOTE	Calls attention to important information, best practices and tips. NOTE is used to address information not related to personal injury, equipment damage, and environment deterioration.

Change History

Issue	Date	Description
01	2019-01-26	This issue is the first official release.

Contents

1 Voice

Voice communication is a method for long distance voice data transmission over networks using various technologies and protocols. Voice communication supports basic voice services, such as fax and modem services, and can be applied in residential as well as enterprise private line services.

Voice services, including POTS, fax, modem, ISDN, and R2 services, apply to multiservice access node (MSAN), fiber to the building (FTTB), fiber to the curb (FTTC), fiber to the home (FTTH), fiber to the office (FTTO), and enterprise private line scenarios.

Access devices support the following basic voice features to help carriers provide high-quality voice services.

SIP value-added services provide more services with easier operations for users and help carriers provide various and flexible services for users. These services improve carriers' competitiveness and user satisfaction.

This topic describes the MGCP protocol and the working principle of MGCP application in VoIP, MoIP and FoIP.

This topic first describes the H.248 protocol, and then describes the protocol mechanism, and last describes the application of H.248 in VoIP, MoIP, and FoIP.

This topic describes the features in relation to the POTS interface, including basic features such as ringing and Z interface and enhanced features.

The integrated services digital network (ISDN) is a CCITT standard, providing integrated transmission service for voice, video, and data. The ISDN enables the voice, video, and data to be transmitted on the data channel simultaneously.

1.11 R2 Access

R2 access enables the MA5600T/MA5603T/MA5608T to be interconnected with a private branch exchange (PBX) through the R2 signaling and helps to provide access services for users over the G.SHDSL ports and E1 ports. As a type of channel associated signaling (CAS), R2 signaling is the international standard signaling based on E1 digital networks.

1.12 FoIP

Fax over Internet Protocol (FoIP) is a fax service provided on an IP network or between an IP network and a traditional PSTN. Fax service is a data service that is widely applied on the PSTN network.

1.13 MoIP

Modem over Internet Protocol (MoIP) is a technology for providing modem services over an IP network or between an IP network and a PSTN network.

1.14 IP Z Interface Extension

IP Z interface extension is that the analog interface between an accsee device and a PBX extends to the remote place through the IP network.

1.15 Key Techniques for Improving Voice Service Quality

Voice service quality is the biggest challenge faced by the IP telephony technology. IP telephony service has a higher requirements on real-time transmission of IP packets. If IP packets are lost, or transmission delay or jitter is introduced due to transmission errors or network congestion, subscribers hear noises during calls, and even more, ongoing calls may be interrupted. The VoIP technology provides a series of technologies, such as codec, echo cancellation (EC), and voice activity detector (VAD) to improve the voice quality.

1.16 Voice Service Maintenance and Diagnosis

The maintenance and diagnosis features of voice services include these features such as the loop-line test, circuit test, call emulation , POTS port loop test, VBD fault diagnosis, Real-time Transport Control Protocol (RTCP) statistics and so on.

1.17 Voice Reliability

This topic describes features related to voice reliability, including dual-homing networking, highly reliable transmission (SCTP), and voice QoS.

1.18 Configuring the VoIP PSTN Service (SIP-based)

The SIP-based VoIP technology makes the transport network evolve to the IP network without decreasing the voice quality, provides more value-added functions for users, and saves expense.

1.19 Configuring the VoIP ISDN BRA Service (SIP-based)

After configuring the SIP interface, you can add VoIP ISDN BRA users on this interface to implement the VoIP ISDN BRA service.

1.20 Configuring the VoIP ISDN PRA Service (SIP-based)

After configuring the SIP interface, you can add VoIP ISDN users on this interface to implement the VoIP ISDN service.

1.21 Configuring the VoIP PSTN Service (H.248-based or MGCP-based)

This topic describes how to configure the VoIP PSTN service when the protocol adopted by the Access node is H.248 or MGCP.

1.22 Configuring the VoIP ISDN BRA Service (H.248-based)

This topic describes how to configure the VoIP ISDN BRA service on an IP network. When the Access node uses the H.248 protocol, the device supports the access of the ISDN BRA user. ISDN technology provides end-to-end (E2E) digital connection and supports multiple types of voice and non-voice telecom services.

1.23 Configuring the VoIP ISDN PRA Service (H.248-based)

This topic describes how to configure the VoIP ISDN PRA service on an IP network. When the Access node uses the H.248 protocol, the device supports the access of the ISDN PRA user. ISDN provides end-to-end (E2E) digital connection and supports multiple types of voice and non-voice telecom services.

1.24 Configuring the R2 Service

With the R2 access technology, the Access node provides access services on common twisted pair cables when interconnecting with the PBX using R2 signaling.

1.25 Configuring the H.248/MGCP-based FoIP Service

This topic describes how to configure the H.248/MGCP-based FoIP service.

1.26 Configuring the SIP-based FoIP Service

This topic describes how to configure the SIP-based FoIP service.

1.27 Configuring the MoIP Service

This topic describes how to configure the H.248/MGCP/SIP-based MoIP service for transmitting the traditional narrowband modem data service over the IP network.

1.28 Adding a POTS IP SPC

A semi-permanent connection (SPC) exclusively occupies a voice channel to meet the communication requirements and to ensure the communication quality for particular and vital access subscribers. To configure an IP SPC, configure the data to set up a direct IP connection between the two ends of the VoIP service. In this manner, the voice media data can be directly transmitted to the peer end.

1.29 Adding a POTS IP SPC Hotline

A semi-permanent connection (SPC) hotline exclusively occupies a voice channel to meet the communication requirements and to ensure the communication quality for particular and vital access subscribers. To configure an IP SPC hotline, configure the data to set up a direct IP connection between the two ends of the VoIP service. In this manner, the voice media data can be directly transmitted to the peer end.

1.30 Configuring the IP Z Interface Extension Service

The following network typically applies to the scenario where Z interface extension private line service needs to be carried over the IP network for the headquarters (HQ) and branch offices of an enterprise after the PSTN network reconstruction. In the following configuration example, the FXO and FXS boards are added for the Z interface extension local MSAN and remote MSAN respectively, board attributes are configured, and IP semi-permanent connections (SPCs) of the IP Z interface extension type are created between the two boards, so that users connected to the FXS board are connected to the corresponding ports on the FXO board through the SPCs.

1.31 Configuring VAGs

The purpose of configuring virtual access gateways (VAGs) is to simulate multiple AGs by using one AG, increasing the usage rate and flexibility of the device.

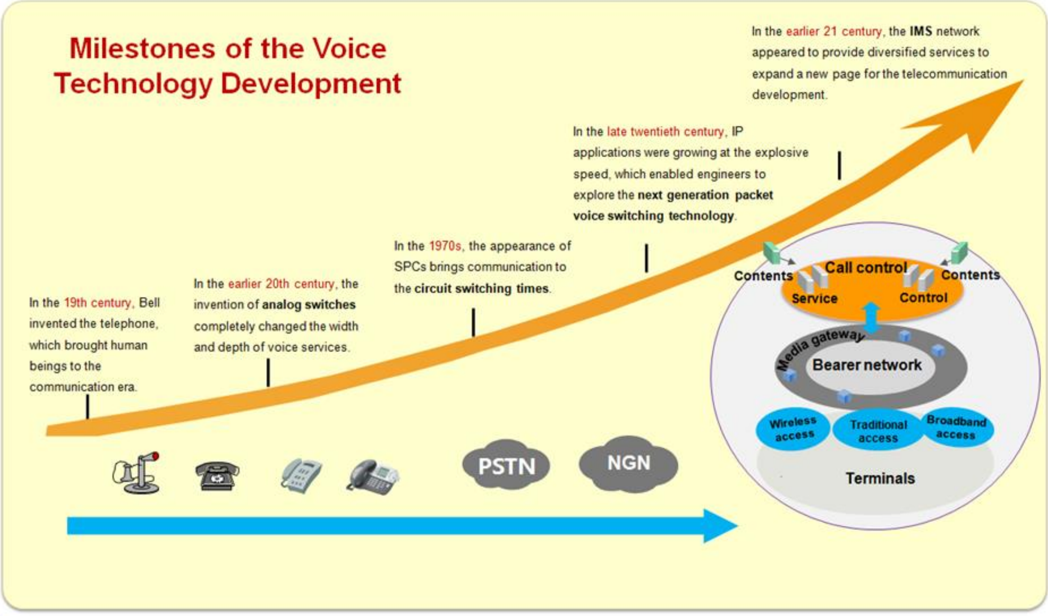
1.32 Configuring the Security and Reliability of the Voice Service

The security configuration of the voice service includes the H.248-based, MGCP-based, or SIP-based device authentication configuration, and the reliability configuration of the voice service includes the dual-homing configuration and the emergency standalone configuration.

1.1 Voice Technology Development

Figure 1-1 shows the voice technology development. The public switched telephone network (PSTN), next generation network (NGN), and IP multimedia subsystem (IMS) are widely used for communication.

Figure 1-1 Voice technology development



Milestones of the Voice Technology Development
In the earlier 21 century, the IMS network appeared to provide diversified services to expand a new page for the telecommunication development.
In the late twentieth century, IP applications were growing at the explosive speed, which enabled engineers to explore the next generation packet voice switching technology.
In the 1970s, the appearance of SPCs brings communication to the circuit switching times.
In the earlier 20th century, the invention of analog switches completely changed the width and depth of voice services.
In the 19th century, Bell invented the telephone, which brought human beings to the communication era.
PSTN
NGN
Contents
Call control
Contents
Service
Control
Media gateway
Bearer network
Wireless access
Traditional access
Broadband access
Terminals

PSTN

PSTN is a connection-oriented network based on timeslot switching. Each circuit connection occupies a timeslot of the pulse code modulation (PCM) basic group, that is, the switching circuit is performed at the rate of 64 kbit/s, which cannot be changed. Advantages and disadvantages of PSTN are as follows:

- Advantages: Fast switching speed, accurate obtaining of call duration, short transmission delay, small jitter, supporting services with high requirements on real-time (especially telephony services)

- Disadvantages: Supporting only 64 kbit/s, exclusively occupying allocated network resources, and low resource usage

NGN

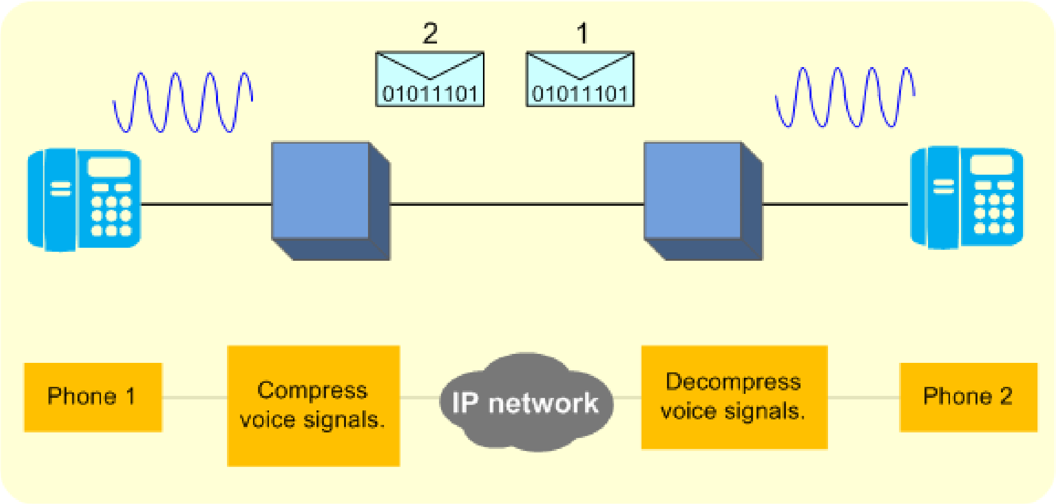
NGN is an integrated network. NGN, a packet-based network, employs the IP technology to build the carrier network and implement the separation of call control from bearing.

For packet voice services, the NGN uses a softswitch as the control layer, IP as the bearer layer, and AG as the access layer. Voice over IP (VoIP) is an important application of packet voice services.

VoIP

To implement VoIP, analog voice signals are compressed and encapsulated into data signals and then transmitted on the IP network, as shown in Figure 1-2. The example usage of VoIP is IP call. In a narrow sense, the VoIP only refers to the voice signal transmission. In a broad sense, the VoIP also refers to data signal transmission, that is, modem over IP (MoIP) and fax over IP (FoIP).

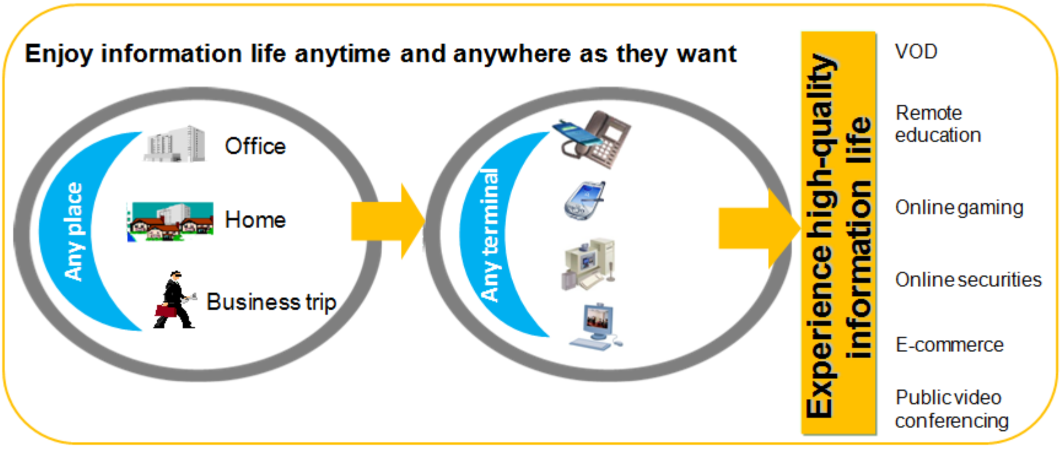
Figure 1-2 VoIP implementation



IMS Network

With the development of telecommunication technology, users want to use voice services to facilitate work and life instead of the traditional voice communication. As a result, a stronger communication platform is required to provide integrated voice, video, and mobility features, as shown in Figure 1-3. The IMS network is developed to meet requirements.

Figure 1-3 Communication requirements


Enjoy information life anytime and anywhere as they want
Any place
Office
Home
Business trip
Any terminal
Experience high-quality information life
VOD
Remote education
Online gaming
Online securities
E-commerce
Public video conferencing

The IMS is a system that controls IP-based multimedia sessions on the NGN. The IMS network contains all core NEs that implement multimedia services, including video, audio, text, and instant messaging (IM).

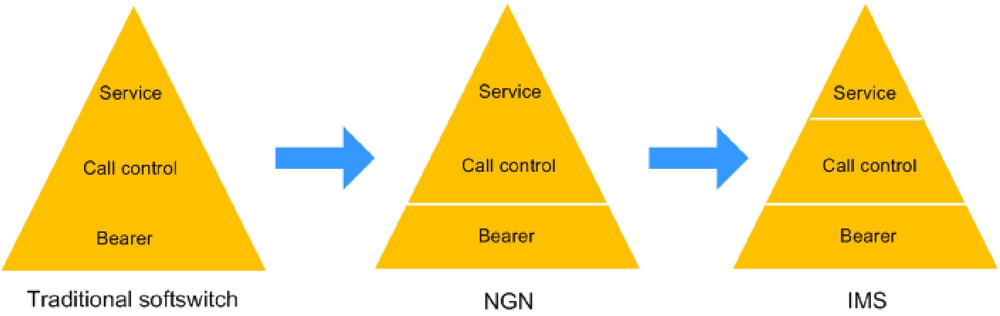
IP =	IP-based transmission
	IP-based session control
	IP-based service implementation
Multimedia =	Supporting multimedia services including video, audio, image, and text
Subsystem =	A system using the advanced network technology and devices

The IMS network was designed by 3rd Generation Partnership Project (3GPP) in the R5 version to support IP-based multimedia services.

The IMS network features the following:

- Same as the softswitch, call control is separated from bearing.
- Services are separated from call control, which speeds up new service provisioning, as shown in Figure 1-4.

Figure 1-4 Control, bearing, and service separation



- The IP-based IMS network is an integrated core network that can be shared by the mobile and fixed networks.
- The IMS network uses E2E SIP signaling. Services and terminals are developed toward intelligence.
  - The control plane using the SIP protocol to control signaling in a centralized manner.
  - The service plane using the SIP protocol to provide a uniformed session mechanism for all services.

1.2 Voice Service Networking Applications

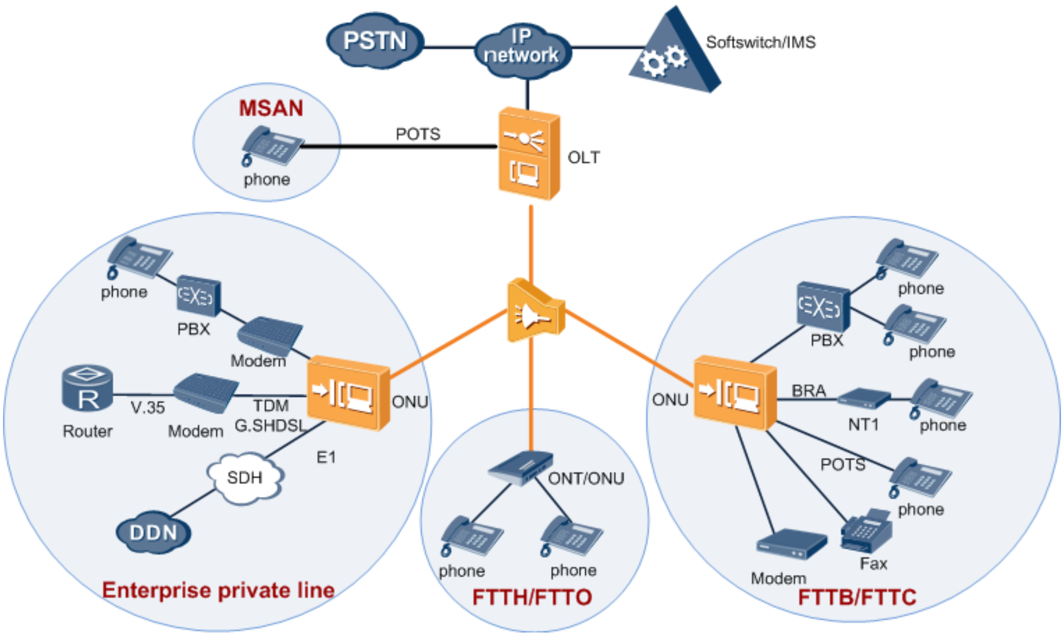
Voice services, including POTS, fax, modem, ISDN, and R2 services, apply to multiservice access node (MSAN), fiber to the building (FTTB), fiber to the curb (FTTC), fiber to the home (FTTH), fiber to the office (FTTO), and enterprise private line scenarios.

Networking Applications

An access gateway (AG) supports the following functions:

- Complies with SIP, H.248, or MGCP, and works with the softswitch or IMS to support the VoIP service.
- Supports ISDN BRA and PRA services.
- Supports the R2 service over E1 lines.
- Supports fax (FoIP) and modem (MoIP) services.
- Supports the TDM SHDSL service using V.35 or E1 upstream transmission, reconstructing traditional voice networks. Compared with the V.35 and E1 services, the TDM SHDSL service supports a longer transmission distance.
  - Prolonged E1 transmission distances: The TDM SHDSL modem on the user side connects to the PBX using an E1 (ISDN PRI) interface, and the modem connects to the AG in TDM G.SHDSL access mode. Then, the AG sends signaling streams to the IP network and exchanges voice service flows with other voice devices using a media gateway (MG).
  - Prolonged V.35 transmission distances: The TDM SHDSL modem on the user side connects to the user-side device using a V.35 (N x 64 kbit/s private line) interface, and the modem connects to the AG in TDM SHDSL access mode. Then, the AG sends data to a DDN network using an SDH network, implementing N x 64 kbit/s DDN private line access.

Figure 1-5 Voice service networking

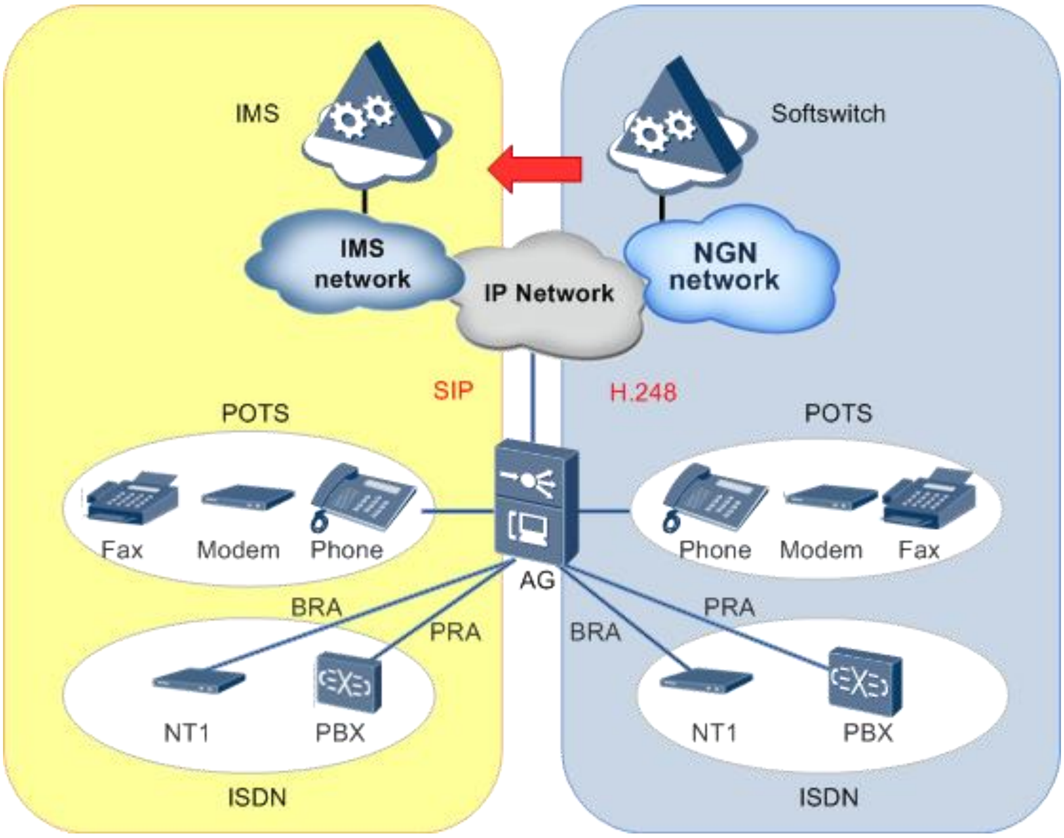


SIP and H.248 Dual-Upstream Transmission

An AG supports both SIP and H.248 upstream transmission. This upstream mode supports smooth migration from a softswitch network to an IP multimedia subsystem (IMS) network, as shown in Figure 1-6. The SIP and H.248 dual-upstream transmission applies to network reconstructions.

- IMS networks provide more value-added services. For the users to be able to migrate from softswitch to IMS, the AG can switch the upstream transmission mode from H.248 to SIP, ensuring that the migration does not interrupt other users' services.
- Each virtual access gateway (VAG) can be configured to support H.248 or SIP.
- The maximum number of users supported by an AG with both H.248 and SIP enabled is the same as that supported by the AG with either of the protocols enabled.

Figure 1-6 Dual-upstream transmission networking



1.3 Voice Feature Overview

Access devices support the following basic voice features to help carriers provide high-quality voice services.

Table 1-1 Basic voice features

Basic Feature	Description
Protocol Voice protocols are used for communication between an access device and an upper-layer gateway control device. The access device supports H.248, Media Gateway Control Protocol (MGCP), and Session Initiation Protocol (SIP). You can run the **display protocol support** command to query the protocol used on the device. If the protocol used on the device is different from the planned protocol, run the **protocol support** command to switch the protocol.	SIP H.248 MGCP
Access Mode	ISDN

Basic Feature	Description
A voice access mode is based on terminal type and service requirements. The access device supports POTS, ISDN, and R2 access modes. Voice communication also supports fax and modem services. - A POTS network is a connection-oriented circuit switched network based on timeslot switching. Each circuit connection uses a timeslot in a pulse code modulation (PCM) group. That is, the circuit switched rate is 64 kbit/s. - The ISDN access can be basic rate access (BRA) or primary rate access (PRA). The BRA access provides 2 B channels and 1 D channel. The rates of B and D channels are 64 kbit/s and 16 kbit/s, respectively. The PRA access provides 30 B channels and 1 D channel. The rates of B and D channels are both 64 kbit/s. - In R2 access mode, the access device connects to a private branch exchange (PBX), which communicates with the access device through R2 signaling.	POTS R2 FoIP MoIP
Key Techniques for Improving Voice Service Quality The VoIP technology provides a series of technologies, such as codec, echo cancellation (EC), and voice activity detector (VAD) to improve the voice quality.	Key Techniques for Improving Voice Service Quality

Table 1-2 Voice service features

Service Type	Description
VoIP The VoIP service uses the IP packet switched network for transmission after the traditional analog voice signals are compressed and packetized, to lower the cost of the voice service.	For the SIP protocol: - SIP-based VoIP Principle - Configuring the VoIP PSTN Service (SIP-based) - Configuring the VoIP BRA Service (SIP-based) - Configuring the VoIP PRA Service (SIP-based) - Configuring the VoIP R2 Service (SIP-based)
	For the H.248 protocol: - H.248-based VoIP Principle

Service Type	Description
	- Configuring the VoIP PSTN Service (H.248-based) - Configuring the VoIP BRA Service (H.248-based) - Configuring the VoIP PRA Service (H.248-based) - Configuring the VoIP R2 Service (H.248-based)
	For the MGCP protocol: - MGCP-based VoIP Principle - Configuring the VoIP PSTN Service (MGCP-based)
MoIP The modem over Internet Protocol (MoIP) refers to the modem service provided on the IP network or between the IP network and the traditional PSTN network.	For the SIP protocol: - SIP-based MoIP Principle - Configuring the MoIP Service (SIP-based)
	For the H.248 protocol: - H.248-based MoIP Principle - Configuring the MoIP Service (H.248-based)
	For the MGCP protocol: - MGCP-based MoIP Principle - Configuring the MoIP Service (MGCP-based)
FoIP The fax over Internet Protocol (FoIP) refers to the fax service provided on the IP network or between the IP network and the traditional PSTN network.	For the SIP protocol: - SIP-based FoIP Principle - Configuring the FoIP Service (SIP-based)
	For the H.248 protocol: - H.248-based FoIP Principle - Configuring the FoIP Service (H.248-based)
	For the MGCP protocol: - MGCP-based FoIP Principle - Configuring the FoIP Service (MGCP-based)
Line hunting Line hunting is a feature that allows a group of ports to share a group of called party numbers by specifying a hunting group and hunting policy. Only the SIP protocol supports this feature.	- Line Hunting Principle - Configuring Line Hunting

Service Type	Description
POTS IP SPC To configure an IP SPC, configure the data (including the local IP address, local UDP port, remote IP address, and remote UDP port), set up an IP direct connection between the two ends of the voice service. In this manner, the voice media data can be directly transmitted to the peer end.	- POTS IP SPC - Adding a POTS IP SPC

Table 1-3 Voice security features

Security	Description
Device authentication Device authentication is a method to improve the security of the core network and prevent illegal devices from registering with the core network device.	For the SIP protocol: - Device Authentication (SIP-based)
	For the H.248 protocol: - Device Authentication (H.248-based)
	For the MGCP protocol: - Device Authentication (MGCP-based)
Dual-homing Dual homing is an NGN (Next Generation Network) total solution. Based on this solution, when the active softswitch or the link from the MG to the active softswitch is faulty, the MG need be switched to the standby softswitch immediately to prevent call services of users connected to the softswitch and the MG from being affected.	For the SIP protocol: - SIP-based Dual-Homing Principle - Configuring the Dual-Homing (SIP-based)
	For the H.248 protocol: - H.248-based Dual-Homing Principle - Configuring the Dual-Homing (H.248-based)
	For the MGCP protocol: - MGCP-based Dual-Homing Principle - Configuring the Dual-Homing (MGCP-based)
Multi-homing As an enhancement of dual-homing, multi-homing is a configuration in which a media gateway (MG) is homed to the primary media gateway controller (MGC), secondary MGC, and disaster-recovery MGC.	- Multi-Homing Principle - Configuring Multi-Homing
Emergency standalone	- Emergency Standalone Principle

Security	Description
Emergency standalone is a solution in which the users on the same MG can call each other even when the interface between the MG and the softswitch is interrupted.	- Configuring Inner Standalone

Table 1-4 POTS port maintenance and test features

Troubleshooting Method	Description
A POTS user loop line test is used to test the electrical indicators of the line from the test device (an access node) to a phone. When users' POTS services are faulty, POTS user loop line tests can be performed to test the performance and electrical indicators of the loop line to diagnose whether the loop line is faulty.	POTS User Loop Line Test
A POTS user circuit test is used to check whether the chip of a POTS board functions normally. If the POTS services are faulty and the loop line works normally, POTS user circuit tests can be used to test the functions (such as the ringing and power feeding) and some parameters (such as the feeding voltage and ringing voltage) of the board circuit to check whether the circuit works normally.	POTS User Circuit Test
A POTS port loop test is used to test the hardware and configurations related to POTS services during device installation or before POTS service provisioning. It helps reduce the number of site visits and minimize maintenance costs.	POTS Port Loop Test
A search tone test is a simple line fault locating function intended for maintenance engineers. In a search tone test, the test module sends voice signals with the specific frequency and amplitude to a line, and then maintenance engineers use a receiver or a dedicated device to detect the signals on the line. In addition, search tone tests can help maintenance engineers pinpoint the specific line among multiple user lines.	Search Tone Test
In a signal tone test, the system sends the signal tone signals to a specific port of a POTS board and makes the port loop back the signals, and then checks whether the loopback signals can be detected. This test function helps maintenance engineers check whether the system can normally process the detection of the user	Signal Tone Test

Troubleshooting Method	Description
off-hook and signal tone and locate hardware faults related to the user off-hook and signal tone playing.	
A call emulation test emulates call functions to verify data configuration for the voice service. The call emulation test can also be used to locate voice service faults.	Call Emulation Test

1.4 Basic Concepts in Voice Services

Learning these basic concepts facilitates deep understanding of voice services.

1.4.1 Voice Media and Signaling

Media and signaling play important roles in voice services.

- Voice media: Used to carry and normally transmit voice communication contents. In the NGN system architecture, Real-Time Transport Protocol (RTP) carries media streams.
- Voice signaling: Used to set up and control voice communication between two telecommunication entities. Different from IP protocols, signaling protocol fields carry commands. Common signaling protocols are MGCP, H.248, and SIP.

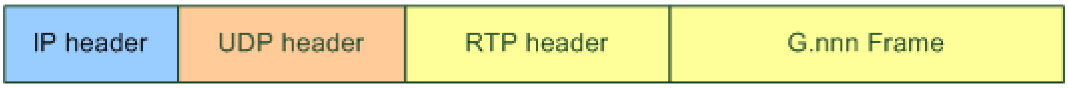
1.4.1.1 Voice Media

RTP

Real-Time Transport Protocol (RTP) is dedicated for multi-media streams over the Internet. In a VoIP network, RTP carries media streams. For details about RTP, see the *RFC3550*.

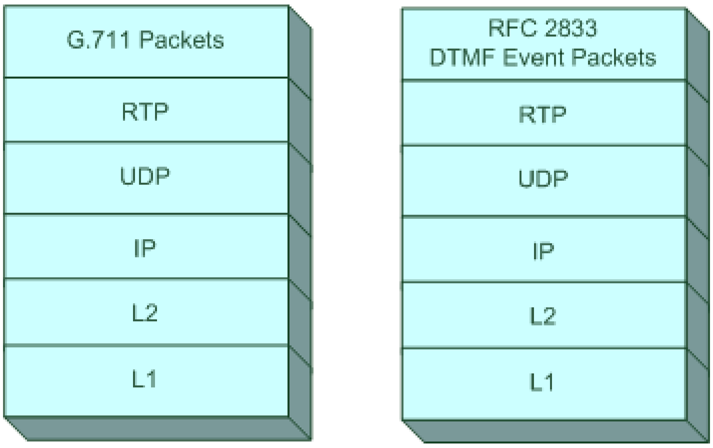
RTP provides end-to-end (E2E) services to transport real time data (including audios and videos) in the format defined in G.711, RFC2833, and RFC2198. Figure 1-7 shows the format of an RTP packet.

Figure 1-7 RTP packet format



RTP runs on top of User Datagram Protocol (UDP) to make use of its multiplexing and checksum services. However, RTP may be used with other suitable underlying network or transport protocols. Figure 1-8 shows the RTP protocol stack model.

Figure 1-8 RTP protocol stack model



RTP receives media streams from the upper-layer device and encapsulates the streams into RTP packets. Then RTP sends the packets to the lower-layer device. The lower-layer protocol transmits RTP and Real-Time Transport Control Protocol (RTCP) packets through different ports. For example, if UDP is used as the lower layer protocol, the protocol uses a port with the ID of an even number to transmit RTP packets and uses the port with the ID of the odd number following the even number to transmit RTCP packets.

RTCP

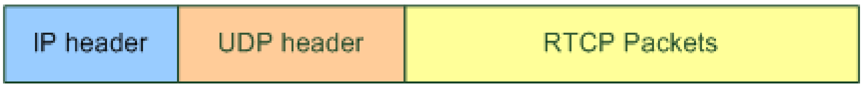
RTP itself ensures real-time data transmission, but cannot provide a mechanism for reliably transmitting data in sequence or a traffic and congestion control mechanism. It provides the mechanisms using RTCP.

RTCP is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets, so that it provides feedback on the quality of the data distribution.

- RTCP packets take 5% of bandwidths.
- RTCP packets contains ring delay, packet loss stream, and jitter for QoS monitoring.

Figure 1-9 shows the format of an RTCP packet.

Figure 1-9 RTCP packet format



RFC 2833

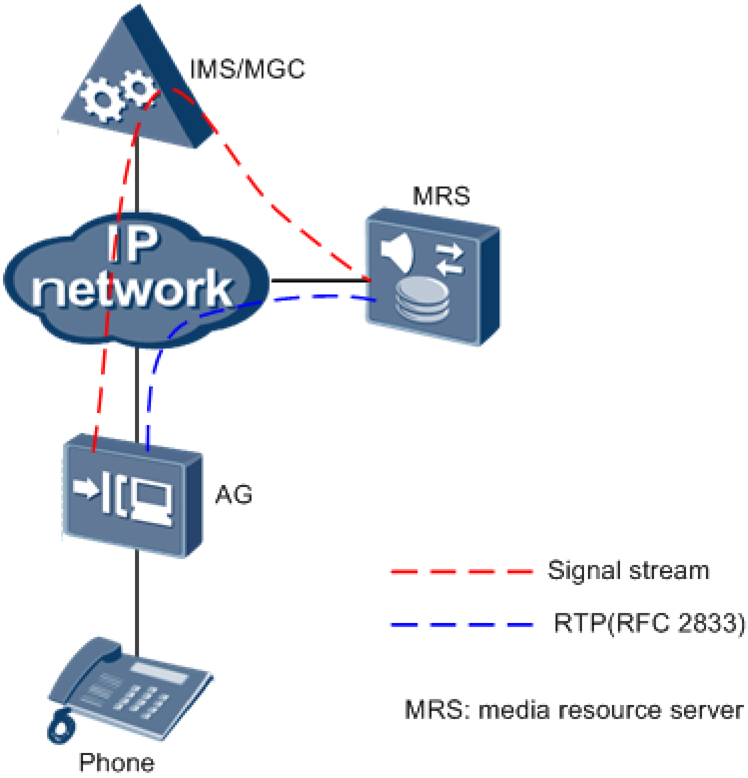
RFC2833 defines a dedicated format to reliably transmit important data, such as signal tone, event, dual tone multi-frequency (DTMF) signal in G.711 and G.729 communication.

Figure 1-10 shows the typical RFC2833 application scenario.

1. After the user dial a number, the softswitch/IMS controls the access gateway (AG) to create and transmit RTP voice media streams to the media resource server (MRS).

2. The MRS plays an announcement of dialing the number to the AG.
3. The user dials the number. Then the number is transmitted to the MRS through the RFC2833 packet that is carried over voice media RTP.
4. The MRS collects and transmits the number to the softswitch/IMS.

Figure 1-10 Typical RFC2833 application scenario



RFC2198 Redundancy

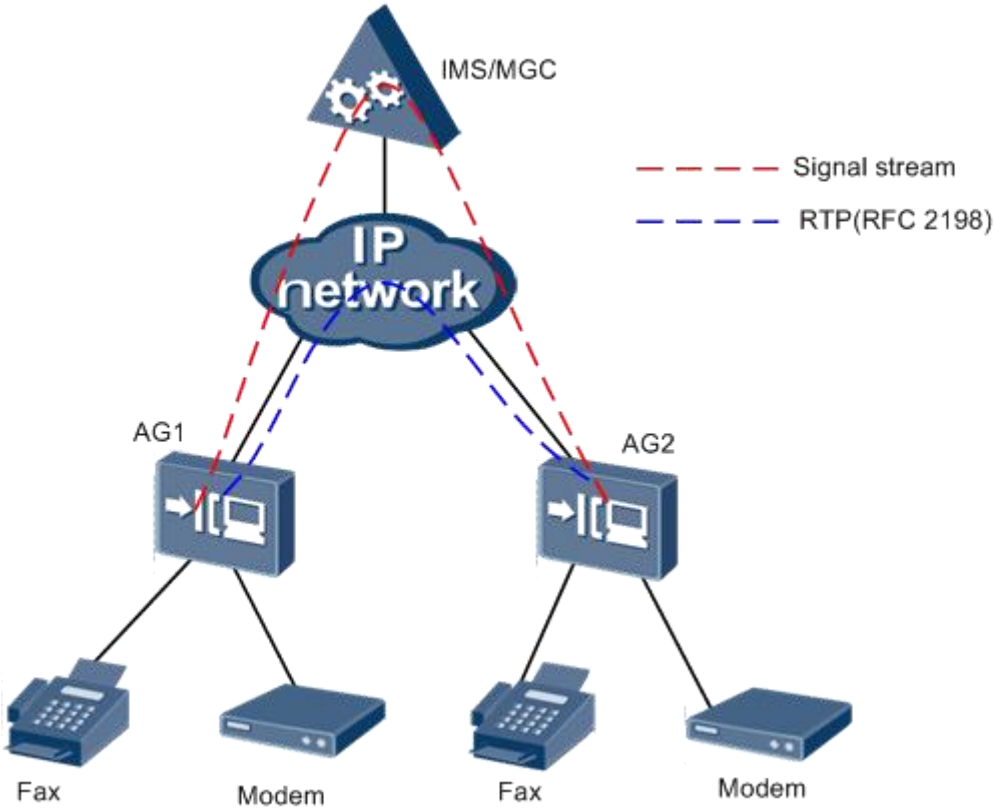
RFC2198 describes the RTP payload format for redundant audio data, which can be used for the RFC2833 digit collecting, fax transparent transmission service, and modem transparent transmission service.

NOTE

RFC2198 redundancy is unnecessary for T.38 fax services, this is because T.38 fax service supports redundancy for its own.

RFC2198 improves the reliability of data transmission through redundant transmission. When the network quality is poor, redundant transmission can ensure the service quality and reduce impacts brought by distorted signals. Figure 1-11 shows RFC2198 redundancy application

Figure 1-11 RFC2198 redundancy application



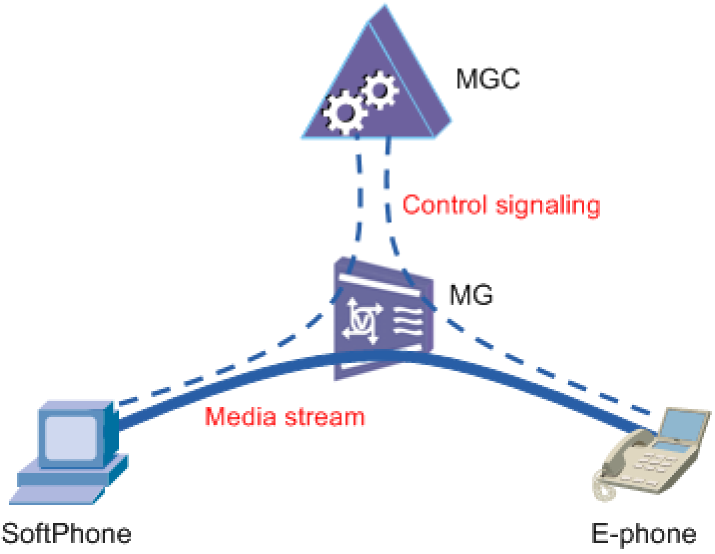
1.4.1.2 Voice Signaling

The signaling technology implements phone calls. The commonly used VoIP control signaling systems contain MGCP, H.248, and SIP.

MGCP Protocol

Media Gateway Control Protocol (MGCP) is defined in the RFC2705 standard and it defines a call control structure in which call control is separated from service bearer. As shown in Figure 1-12, call control is separate from the media gateway (MG) and is processed by the media gateway controller (MGC). Therefore, MGCP is in nature a master-slave protocol. The MG establishes various service connections under the control of the MGC.

Figure 1-12 MGCP Master-Slave Control



H.248 Protocol

H.248 is the same type of protocol as MeGaCo and completed by the ITU-T and IETF together, used as a media gateway control protocol between an MGC and an MG. It takes the place of MGCP. H.248 features the following:

- Functions on the basis of MGCP and therefore it inherits all advantages of MGCP.
- Works in master-slave mode.
- Uses binary coding or text coding for H.248 messages. MGC must support these two coding modes and MG supports either of them.
- Uses User Datagram Protocol (UDP), Transmission Control Protocol (TCP) or Stream Control Transmission Protocol (SCTP) (IP-based signaling transmission) for underlying transmission.

Compared with MGCP, H.248 has the following advantages:

- Supports voice services and multi-media connections.
- Supports text coding and binary coding.
- Features expandability.

SIP Protocol

Session initiation (SIP) is a session control protocol running at the application layer, which sets up, modifies, and terminates a session. A session can be an application, such as multi-media conference and Internet call.

Comparison Between the H.248 Protocol and SIP Protocol

Table 1-5 describes comparison between the H.248 protocol and SIP protocol.

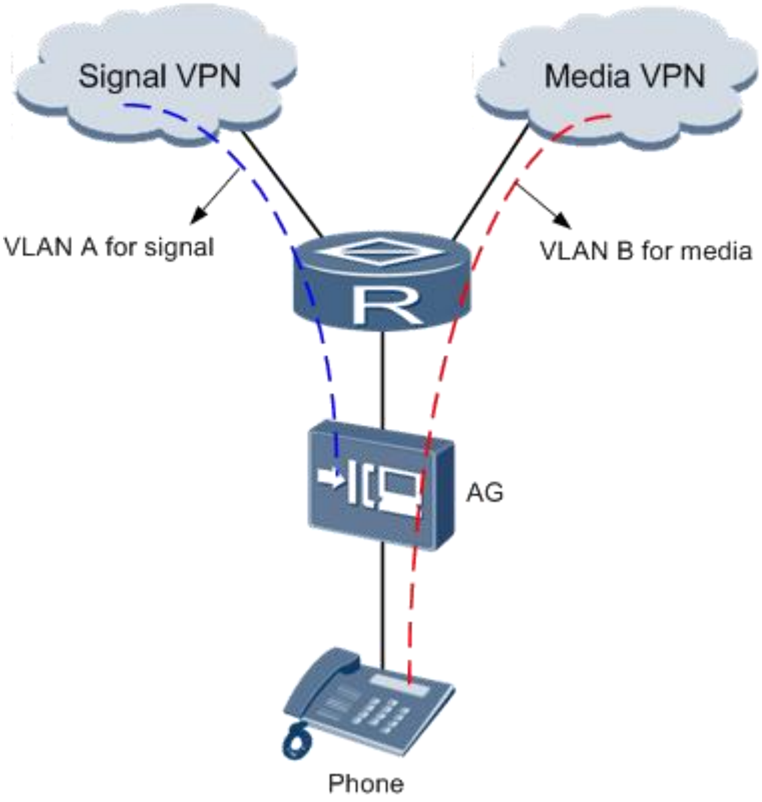
Table 1-5 Comparison Between the H.248 protocol and SIP protocol

Item	SIP Protocol	H.248 Protocol
Standard	IETF/TISPAN	ITU_T/TISPAN
Architecture	Distributed/Intelligent clients	Centralized/Dumb end_point
Call control	Proxy/Redirect Server	Call agent/MGC
Transport protocol	UDP/TCP/SCTP	UDP/TCP/SCTP
Multi-media service supported	Yes	Yes
Supplementary service	Provided by endpoints or by call control	Provided by call control
ISDN service	Not defined in TISPAN R1	Use IUA Support

1.4.1.3 Separation of Media and Signaling Streams

Separation of media and signaling streams indicates that signaling streams (H.248/SIP) and media streams can be transmitted upstream to different virtual private networks (VPNs) through different IP addresses and VLANs. The separation facilitates network planning and meets control requirements, as shown in Figure 1-13.

Figure 1-13 Separation of media and signaling streams



Application scenarios of separation of media and signaling streams:

- Signaling and media flows are transmitted upstream to different VPNs and they use different control policies, such as QoS.
- Different user groups use different signaling IP addresses/VLANs and media IP addresses/VLANs. This meets the special monitoring requirements.

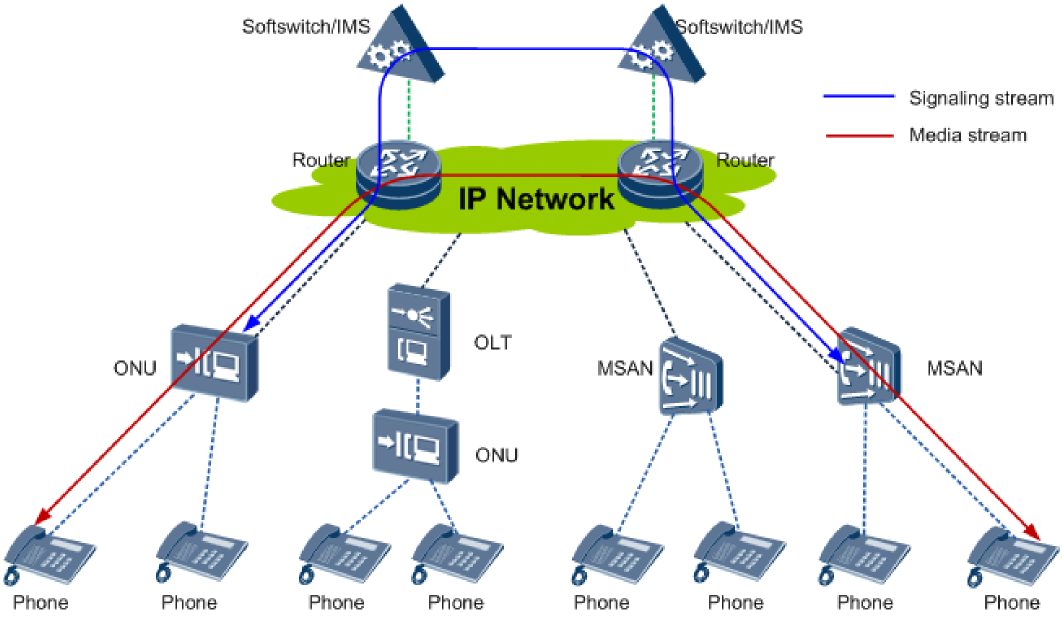
1.4.1.4 External Direction of Voice Media Streams and Signaling Streams

Voice services can be classified into the following 3 scenarios in which media streams and signaling streams are transmitted in different directions.

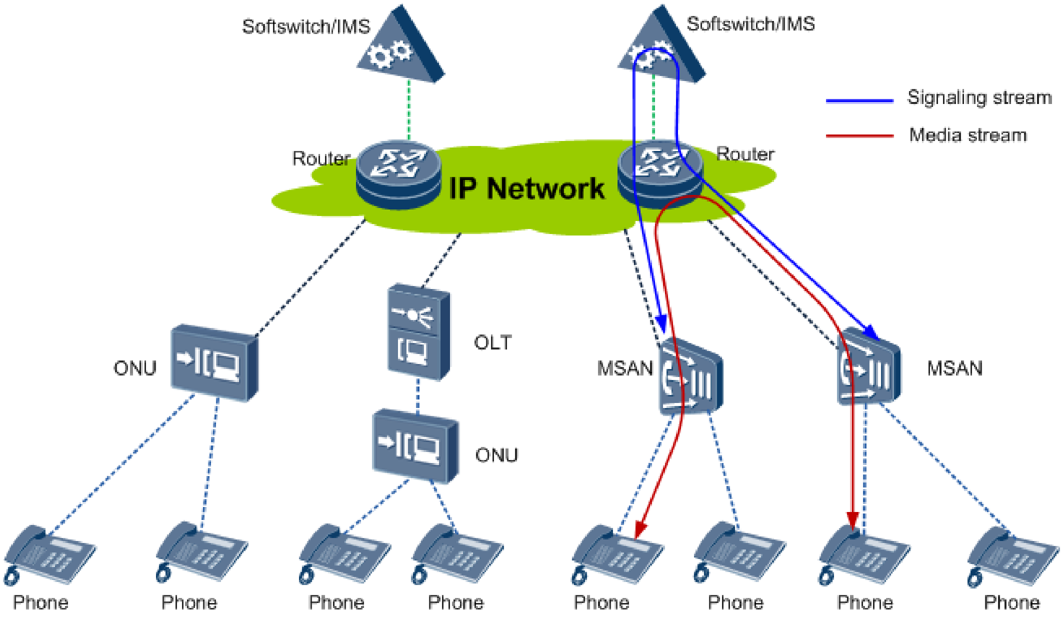
- Communication under different softswitches/IMS
- Communication under the same softswitch/IMS but different access gateways (AGs)
- Communication under the same softswitch/IMS and same AG

The following shows directions of voice media streams and signaling streams in these 3 scenarios:

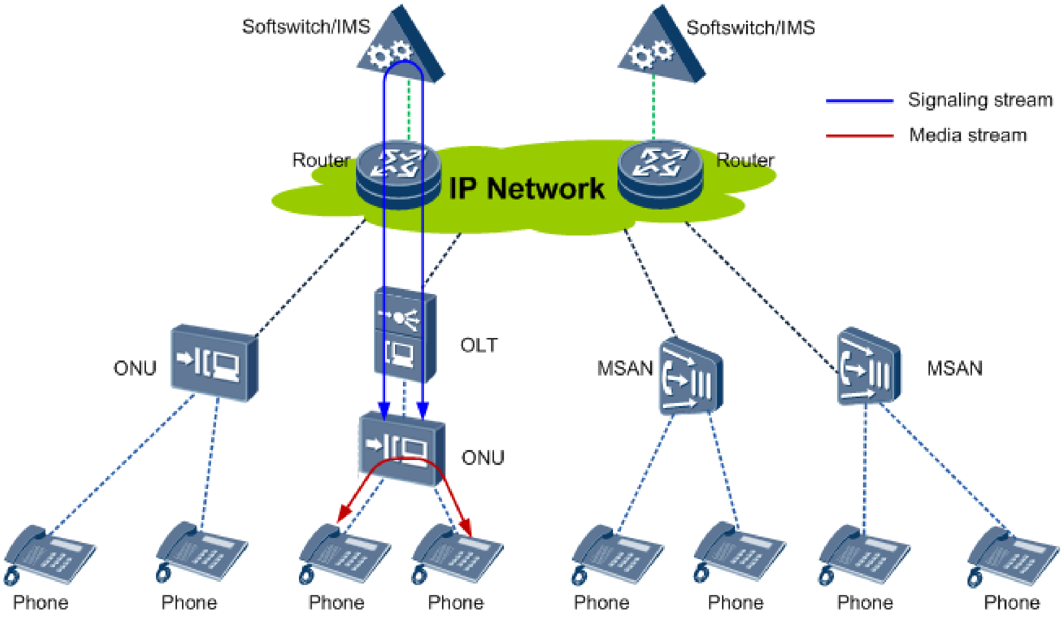
Communication Under Different Softswitches/IMS



Communication Under the Same Softswitch/IMS but Different AGs



Communication Under the Same Softswitch/IMS and Same AG



1.4.1.5 Internal Direction of Voice Media Streams and Signaling Streams

POTS Access Mode

The following flash video demonstrates the directions of media streams and signaling streams inside an AG device in POTS access mode.

Functions of each module are as follows:

- The subscriber line interface circuit (SLIC) processes analog signals, including feeding phone sets, sending voice frequencies to phone sets, and generating ringing, as well as detecting off-hook, pulse dialing, on-hook, and hookflash signals.
- The coder/decoder (CODEC) converts between analog and digital signals. It converts analog signals to digital signals in the upstream direction and digital signals to analog signals in the downstream direction.
- The digital signal processor (DSP) supports the following functions:
  - Codes and decodes voice signals. The DSP encapsulates the digital signals sent by the CODEC into VoIP packets in the upstream direction and restores the VoIP media streams transmitted over the GE bus to digital signals in the downstream direction.
  - Manages the SLIC and CODEC using the SPI bus.
- The GE LSW and voice processing module process signaling streams and media streams. They determine whether to discard or forward signaling/media streams based on the IP address of the signaling/media streams and the UDP port number. The voice processing module is a daughter board on the control board.

ISDN Access Mode

The following flash video demonstrates the directions of media streams and signaling streams inside an AG device in ISDN access mode.

Functions of each module are as follows:

- Network termination (NT) 1: functions similarly as the physical layer of the OSI reference model. The functions of NT1 are associated with inherent physical and electrical characteristics of the network.
- Terminal adapter (TA): Non-ISDN terminals do not support channel-D functions. To connect a non-ISDN terminal to the ISDN network, a TA is required.
- ISDN chip group: connects to NT1 using a U interface for ISDN BRA services. The ISDN chip group converts analog signals to digital signals and implements codec of a U interface (such as 2B1Q and 4B3T codes).
- E1 chip group: connects the E1 interface to NT 1 for the ISDN PRA service. It converts analog signals to digital signals.
- DSP: codes and decodes voice signals. The DSP encapsulates the digital signals sent by the ISDN or E1 chip group into VoIP packets in the upstream direction and restores the VoIP media streams transmitted over the GE bus to digital signals in the downstream direction.
- The GE LSW and voice processing module forward signaling streams and media streams. They determine whether to discard or forward signaling/media streams based on the IP address of the signaling/media streams and the UDP port number. The voice processing module is a daughter board on the control board.

1.4.2 VAG

The virtual access gateway (VAG) is a solution in which one AG device can be simulated into multiple AG devices. Different VAGs can connect to different IMSs/softswitches, as shown in Figure 1-14. The purposes of the VAG are as follows:

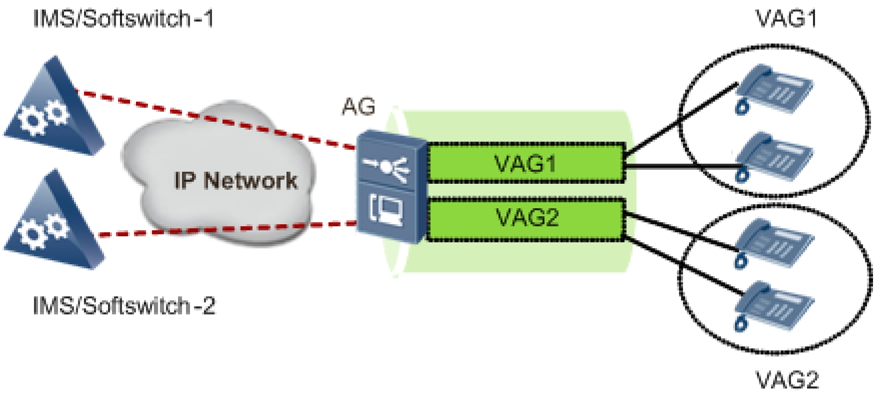
- Differentiated service

  Simulate multiple logical AGs on a physical AG, and enable different logical AGs to connect to different customer groups. In this way, different convergence ratios(Convergence ratio indicates the ratio between the sum of DSP channel resources and DSP resources concurrently occupied by the supported users.) are provided for different customer groups to meet the identified service requirements.
- Virtual operation

  In the AG networking application, certain small-scale operators do not buy independent AG devices, but rent the AG devices from other operators to provide the service. For the renters, they may rent out a device to multiple operators to improve the device utilization rate. Simulate multiple logical AGs on a physical AG, and enable different logical AGs to connect to different softswitches. In this way, the wholesale service requirements can be met.
- Sharing load on the IMS/softswitch

  One IMS/softswitch has limited capacity. If subscribers connected to different VAGs are distributed to multiple IMS/softswitches, the workload on one IMS/softswitch becomes lighter.

Figure 1-14 VAG networking



The VAG call process and the traditional call process are similar. The main differences are as follows:

- The DSP distributes resources. After a subscriber picks up the phone, the AG applies for DSP resources. If DSP resources are assigned to the VAG, the VAG obtains DSP resources from the shared resource pool. If DSP resources are not assigned to the VAG, the VAG obtains DSP resources from the exclusive resource pool.
- The IP addresses in the IP packets during the connections and conversations of the subscribers under the same VAG are the media IP address corresponding to the VAG.

Each VAG can be configured and managed independently. For the softswitch, each VAG is an independent AG, and can be configured with related attributes separately, such as authentication, ringing mapping, and terminal layered mode. When configuring a subscriber, specify a VAG for the subscriber. Subscribers under different VAGs can share the same terminal ID. The total number of subscribers configured on all VAGs cannot exceed the maximum number configured in the system.

To configure a VAG, add a VAG interface (MG interface), and enter the VAG interface mode to configure interface parameters by running commands. Each VAG can be configured with independent signaling IP address and media stream IP address (the signaling IP address and media stream IP address in the same VAG can be different) in different VLANs.

1.4.3 Local Digitmap

A digitmap is a dialing scheme configured on an AG. The AG collects digits dialed by calling parties based on the digitmap. With the digitmap, the AG reports a group of digits each time, reducing the number of signaling exchanges between the AG and the softswitch/IMS, thereby improving efficiency. The digitmap configured on the AG is called local digitmap. An H.248-compliant AG can use the local digitmap or digitmap issued by the softswitch. A SIP-compliant AG must use the local digitmap

Definition

After a subscriber dials a number, the AG matches digits of the number. If the matching is successful, the AG sends the collected digits to the softswitch/IMS. The softswitch/IMS analyzes the number and corresponding services. If the matching fails, the AG discards the digits.

On the NGN, terminals send called numbers to the softswitch/IMS on the control layer through the AG on the access layer. Numbers can be transmitted in in-band or out-band mode.

- In-band transmission occupies IP media stream resources. In this mode, the AG transfers numbers in RTP or RFC 2833 format to the upper layer network. Such processing is not related to digitmaps.
- In out-band transmission mode, the AG transfers numbers through signaling streams. The out-band transmission has the following two modes:
  - Reporting a number digit by digit: The AG reports a digit to the softswitch/IMS after a subscriber dials a digit. The number of signaling exchanges between the AG and softswitch/IMS is determined by the number of digits contained in a dialed number. Such processing is not related to digitmaps.
  - Reporting a complete number: The AG reports a complete number dialed by the subscriber to the softswitch/IMS using one signaling message. This type of processing lowers the burden on the softswitch/IMS and is preferentially selected. To report a complete number, the AG must know when to collect digits and when to report digits. The digitmap determines whether dialed digits are valid.

Digitmap Example

The following digitmap is used as an example:

[2-8]xxxxxxx | 13xxxxxxxxx | 0xxxxxxxxx | 9xxxx | 1[0124-9]x | * | # | x.# | [0-9*#].T

This digitmap consists of nine character strings.

- The first character string "[2-8]xxxxxxx" indicates that the matched number must contain eight digits, the first digit must be one of 2 to 8, and remaining seven digits can be any of 0 to 9.
- The second character string "13xxxxxxxxx" indicates that the matched number must contain 11 digits, the first digit must be 1, the second digit must be 3, and remaining nine digits can be any of 0 to 9.
- The third character string "0xxxxxxxxx" indicates that the matched number must start with 0 and contain 10 digits.
- The fourth character string "9xxxx" indicates that the matched number must start with 9 and contain 5 digits.
- The fifth character string "1[0124-9]x" indicates that the first digit must be 1, the second digit cannot be 3, and the third digit can be any of 0 to 9.
- The sixth character asterisk (*) indicates that the matched character must be an asterisk (*).
- The seventh character pound sign (#) indicates that the matched character must be a pound sign (#).
- The eighth character string "x.#" indicates that the AG matches one of 0 to 9 for multiple times or even does not match any digits and stops the matching until detecting a pound sign (#).
- The ninth character string "[0-9*#].T" indicates that the AG starts a timer, matches digits 0 to 9 for multiple times or even does not match any digits, asterisk (*), or pound sign (#), and stops the matching only after the timer expires.

NOTE

The dot (.) indicates zero or multiple times of matching. However, at least one character must be matched for this character string. Therefore, at least one of 0-9, *, and # must be matched.

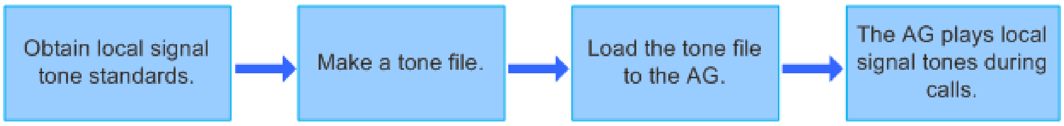
1.4.4 Local Tone

The softswitch/IMS or AG can play tones to terminals. Tones played by the AG are called local tones. Before enabling the AG to play local tones, load a tone file to the AG. For parameter tones, configure them on the AG.

Tone File

Signal tone standards vary depending on country requirements. As shown in Figure 1-15, customize a tone file, load the tone file to the AG, and the AG plays the customized tones, such as dial tone, ring back tone, and busy tone during calls.

Figure 1-15 Tone played by the AG using the tone file



Obtain local signal tone standards.
Make a tone file.
Load the tone file to the AG.
The AG plays local signal tones during calls.

Voice files, generally named **voice.efs**, are stored in the flash memory of a control board. A voice file describes the tone playing types supported by the DSP. The description covers the signal tone type, frequency, duration, and level. After the system is initialized, tone playing parameters are configured on the DSP. When the system requests to play a tone for a subscriber, the DSP reads the configuration and generates the desired signal tone on a real-time basis.

The signal tones specified in a voice file can be parameter tones, waveform tones, or announcements.

- Parameter tones are simple tones, including dialing tones, busy tones, and ring back tones. The system issues parameter tone attributes, such as the frequency, energy, duration, and cadence, to the DSP. The DSP then generates parameter tones.
- Waveform tones are simple tones. The system records these tones into a voice file and stores the file in the flash memory of a control board. Whenever a board starts up, the voice file is loaded to board DSP. When the system needs to play a waveform tone for a subscriber, the DSP plays the recorded voice data for the subscriber.
- Announcements are audio messages played to subscribers, such as "The subscriber you dialed is busy. Please call later." The system records announcements into a voice file and stores the file in the flash memory of a control board. Whenever a board starts up, the voice file is loaded to board DSP. When the system needs to play an announcement for a subscriber, the logic or DSP plays the recorded voice data for the subscriber.

In addition to voice file recording, the system supports parameter tone customization. A customized parameter tone takes effect in the next tone playing, which does not require service board resetting.

Customized Parameter Tones

Users can customize a parameter tone by specifying a series of cadences and tone playing rules. Each cadence defines the signal frequency, energy, and duration.

In a parameter tone, cadences can be played in one of the following ways:

- Sequential play: Each cadence is played once in sequence according to the break-make ratio.

- Continuous play: All cadences are cyclically and continuously played according to the break-make ratio.
- Cyclic play: All cadences are cyclically played and the number of cycles can be defined.

NOTE

In a parameter tone, continuous cadences can be cyclically played. For example, in Figure 1-16, special dial tone cadences 1 and 2 can be continuously played. When configuring a cadence cycle, ensure that the following requirements are met:

- The cycle end cadence must be greater than the cycle start cadence. The start and end cadences must have been configured.
- If multiple cycles are configured, the start cadence of the next cycle must be greater than or equal to the end cadence of the previous cycle. In addition, the cadences in each cycle cannot overlap or nest. For example, a parameter tone contains eight cadences, cadence 1 through cadence 8, and cycle 1 contains cadences 1, 2, and 3. Then, the start cadence of cycle 2 must be greater than or equal to cadence 3 and cannot contain cadence 1 or cadence 2.

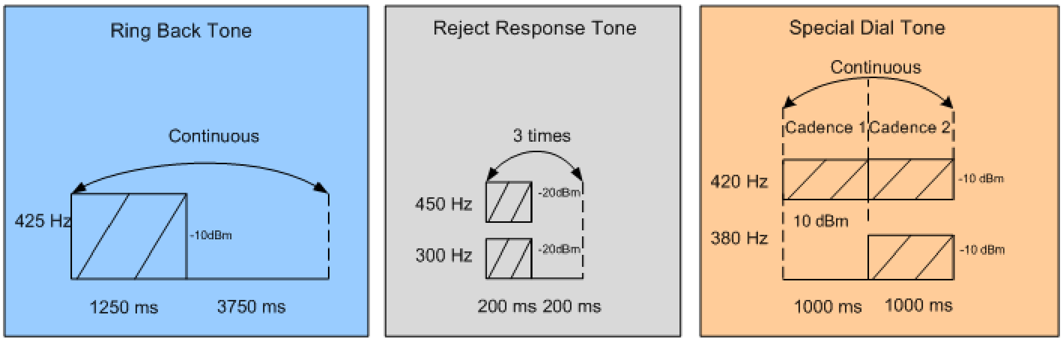
A cadence contains the following items:

- Frequency: A cadence can contain only one frequency, two frequencies, three frequencies, or four frequencies.
- Energy: The energy obtained when the POTS gain is 0 dBm and the line length is 0 km is used in this section.
- Play mode: A cadence can be played only once or repeatedly played according to the break-make ratio. If the cadence is configured to repeatedly play, the number of plays is not limited.

Figure 1-16 shows ring back tone, reject response tone, and special dial tone diagrams.

- A ring back tone is continuously played with a single cadence. The cadence contains one frequency, which is 425 Hz. The energy is -10 dBm. The cadence is played according to the break-make ratio. That is, the cadence is played for 1250 ms and stopped for 3750 ms.
- The reject response tone is cyclically played with a single cadence. The number of cycles is 3. The cadence contains two frequencies, which are 300 Hz and 450 Hz, respectively. The energy is both -20 dBm. The cadence is played according to the break-make ratio. That is, the cadence is played for 200 ms and stopped for 200 ms.
- The special dial tone is continuously played with two cadences, cadence 1 and cadence 2.
  - Cadence 1 contains one frequency, which is 420 Hz. The energy is -10 dBm. The cadence is played only once and the duration is 1000 ms.
  - Cadence 2 contains two frequencies, which are 380 Hz and 420 Hz, respectively. The energy is both -10 dBm. The cadence is played only once and the duration is 1000 ms.

Figure 1-16 Example of generating parameter tones



Feature Dependencies and Limitations

- Users only can customize parameter tones. If the AG is required to play a waveform tone or announcement, a voice file is required.
- If a parameter tone is both customized and configured using a voice file, the customized configuration preferentially takes effect.
- After the voice file is loaded or parameter tones are configured on the AG, and the softswitch or IMS issues tones, tones issued by the softswitch or IMS take effect preferentially. If the softswitch or IMS does not issue tones, the voice file loaded or parameter tones configured take effect preferentially.

1.4.5 Accounting

This section describes accounting on pay phones using coins or IC cards.

Introduction

The AG supports three types of accounting: polarity reversal accounting, 12/16KC accounting, and polarity reversal pulse accounting.

- Polarity reversal accounting: The pay phone starts to account immediately after detecting a voltage reversal between A and B wires.
- 12/16KC accounting: It is also called KC accounting. The pay phone performs accounting based on 12 kHz or 16 kHz high frequency signals sent by the POTS board.
- Polarity-reversal pulse accounting: The pay phone performs accounting based on standard pulse signals sent by the POTS board.

Polarity Reversal Accounting

Polarity reversal refers to the voltage reversal between wires. For example, if the voltage between A and B wires is Vtp, the reversed voltage is -Vtp. After detecting a polarity reversal signal, the pay phone starts to account. The pay phone stops accounting when the voltage between A and B wires changes from -Vtp to Vtp, as shown in Figure 1-17. The polarity reversal is classified into hard polarity reversal and soft polarity reversal.

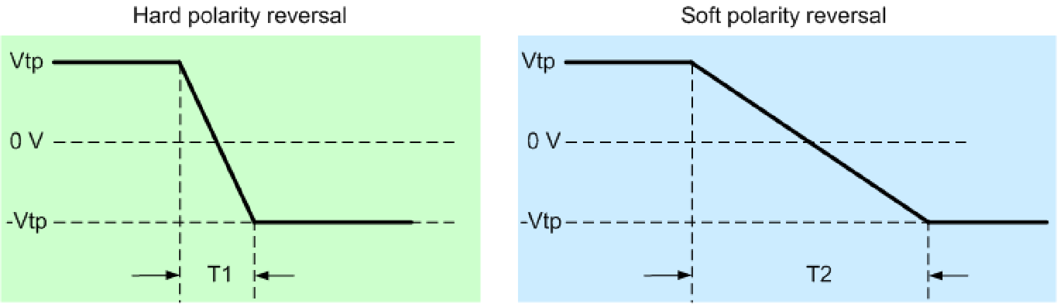
- Hard polarity reversal: also called quick polarity reversal. Specifically, the voltage is reversed in a short period of time, shorter than 3 ms in general. The hard polarity reversal brings great interference on lines.

- Soft polarity reversal: also called slow polarity reversal. Specifically, the voltage is reversed in a long period of time, longer than 80 ms in general. The soft polarity reversal brings small interference on lines.

NOTE

Some terminals are faulty if the polarity reversal time is long. In this situation, the hard polarity reversal must be used, although it brings great interference on lines.

Figure 1-17 Hard and soft polarity reversal



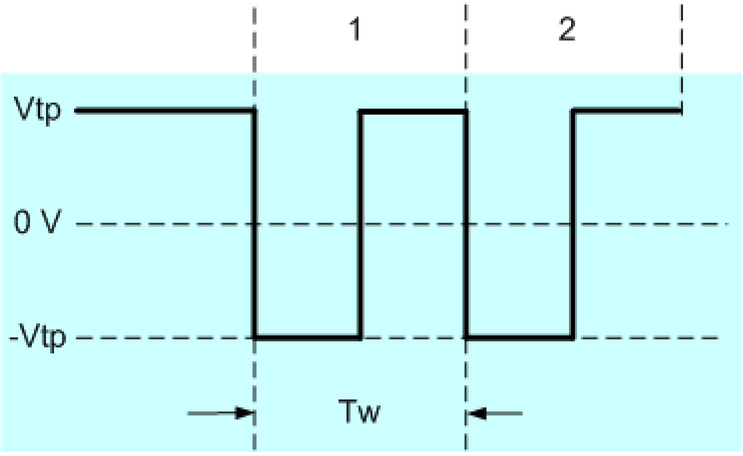
12/16KC Accounting

When detecting that both the calling and called parties enter a session, the softswitch/IMS requests the AG to send 12 kHz or 16 kHz high frequency pulse signals to the pay phone. The pay phone performs the accounting once after detecting such a pulse signal. For example, if the carrier charges subscriber 1 cent for one-minute call, the softswitch/IMS sends such a pulse signal at the interval of one minute. If the pay phone receives three such pulse signals after the call ends, the subscriber is charged 3 cents. After detecting that the call ends, the softswitch/IMS requests the AG to stop sending such pulse signals to the AG.

Polarity Reversal Pulse Accounting

The principle of the polarity reversal pulse accounting is the same as that of the polarity reversal accounting. The difference lies in that the POTS board sends standard pulse signals to pay phones. As shown in Figure 1-18, the pulse width is Tw and there are a total of two pulse signals.

Figure 1-18 Polarity reversal pulse accounting



1.4.6 Hookflash

Definition

When a phone is in the offhook state, the terminal generates an onhook signal for a period.

Action

To generate hookflash, you can quickly press the hookflash button or **R** (generally) on an ordinary phone.

Application

During a call, the user wants to start some new services. For example, the call forwarding service: user A calls user B. During the call, user B presses the hookflash button and hears the special dial tone. User B dials the number of user C and communicates with user C. User B hangs up the phone, and then user A communicates with user C.

Hookflash Signal

Hookflash is short-time phone onhook actually. However, onhook signals last only a short period of time and therefore hookflash cannot be determined as phone onhook. Both upper and lower thresholds for hookflash are set in the system. When the last duration of an onhook signal is within the range between the upper and lower thresholds, the onhook signal is determined as the hookflash signal.

Upper and lower thresholds for hookflash are defined differently in countries. For example, in China, the upper threshold is 350 ms and lower threshold is 100 ms. This indicates that if an onhook signal lasts 100-350 ms, it is determined as a hookflash signal.

1.4.7 Dual Tone Multi Frequency

Introduction

DTMF means that the tones of two frequencies are overlaid to represent a number, as shown in Table 1-6.

Table 1-6 Mapping between frequencies and numbers

Unit: Hz	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

When numbers are dialed on the phone, the dialed numbers are converted into the dual-frequency overlay tones. The DSP detects the dialed numbers by checking the DTMF.

The supported DTMF-specific functions are as follows:

- DTMF erasure: After the DSP detects DTMF signals, it erases the DTMF signals from the RTP media stream.
- DTMF transparent transmission: After the DSP detects DTMF signals, it retains the DTMF signals in the RTP media stream.
- DTMF RFC2833 transmission: After the DSP detects DTMF signals, it erases the DTMF signals from the RTP media stream and sends the DTMF information in RFC2833 transmission mode.

Reference Standards and Protocols

ITU-T Q.24

1.4.8 Calling Indication

In a voice call, common calling indications (CINDs) are as follows:

- Call attempt per second (CAPS): Used to measure the volume of concurrent calls.
- Busy hour call attempts (BHCA): Used to measure the system capability of processing calls. BHCA = CAPS/3600.
- Erlang (ERL): Used to measure the traffic. Erl = Calls per hour x Average call hold duration (unit: s)/3600.
- Call loss count: Indicates failed calls.
- Convergence ratio: Used to measure the ratio of system trunk capacity to user capacity.

1.5 SIP Voice Feature

1.5.1 What Is the SIP Protocol

Definition

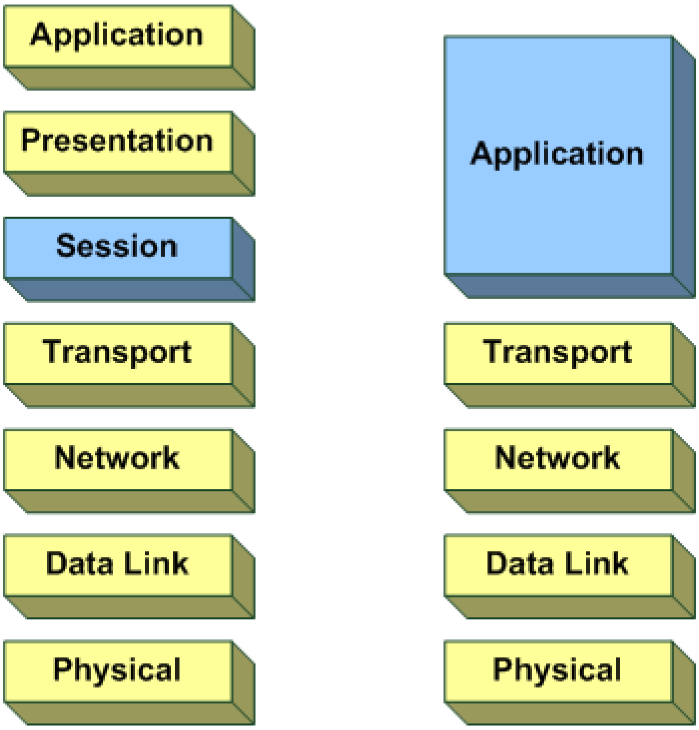
Session Initiation Protocol (SIP), defined in RFC 3261, is used for setting up, modifying, and terminating sessions with one or more participants. The session can be a multimedia meeting, distance learning, or Internet telephony. SIP can be used for initiating sessions or inviting a member to join a session that has been set up otherwise. SIP transparently supports the mapping of names and the redirecting service, which facilitates the implementation of intelligent network, and personal mobile service. Once the session is set up, media streams are simply transmitted at the bearer layer through the Real-time Transport Protocol (RTP).

Position of the SIP Protocol on the Network

The SIP protocol is a signaling control protocol at the application layer. In the five-layer TCP/IP model, SIP is an application layer protocol. In the seven-layer OSI model, SIP is a session layer protocol. Figure 1-19 shows the position of the SIP protocol on the network.

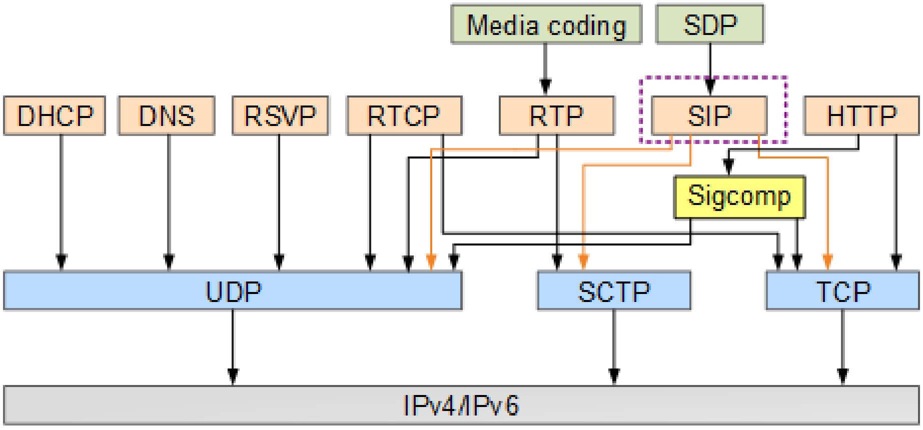
The SIP protocol is independent from transmission protocols but is carried on different transmission protocols, such as, UDP, TCP, TLS, and SCTP. SIP is always carried on the UDP for its efficient transmission.

Figure 1-19 Position of the SIP protocol on the network



The SIP protocol must work together with other protocols to complete multimedia calls. Figure 1-20 shows positions of the SIP and other protocols.

Figure 1-20 Positions of the SIP and other protocols



The SIP protocol works with the Real-time Transmit Protocol (RTP), Real-Time Transport Control Protocol (RTCP), domain name server (DNS), Resource ReServation Protocol (RSVP), and Session Description Protocol (SDP) to complete multimedia calls.

- RTP: A protocol defined by RFC 3550 for transmitting E2E real-time data. It provides the following functions for a series of E2E real-time data transmission services: payload type identification, sequence number arranging, timestamp, and transmission monitoring.
- RTCP: A protocol that controls transmission of real-time media streams.
- RSVP: A protocol that preserves network resources.
- SDP: A text-based application layer protocol for describing multimedia sessions.
- Sigcomp: A mechanism defined by RFC 3320 and used by application layer protocols to compress messages before they are sent to the network.

Advantages of the SIP Protocol

SIP will revolutionize the mode of communication service provisioning and the users' habit of communication consumption. An innovating communication mode integrating video phone service, messaging, Web service, e-mail, synchronous browsing, and conference call will be introduced to the telecommunication industry. Adopting SIP as the control layer protocol has the following advantages:

1. Based on an open Internet standard, SIP has inherent benefits in the integration and interoperability of voice and data services. SIP can implement across-media and across-device call control, and supports various media formats. SIP also supports dynamic adding and deleting of media streams, which make it easier to support richer service features.
2. SIP is intelligently extensible to the service and terminal side, reducing the network load and facilitating the provisioning of service.
3. SIP supports mobile functions at the application layer, including the dynamic registering mechanism, location management mechanism, and redirecting mechanism.
4. SIP supports features such as presence, fork, and subscription, which facilitates development of new services.
5. As a simple protocol, SIP has generally acknowledged extensibility.

1.5.2 Mechanism of the SIP Protocol

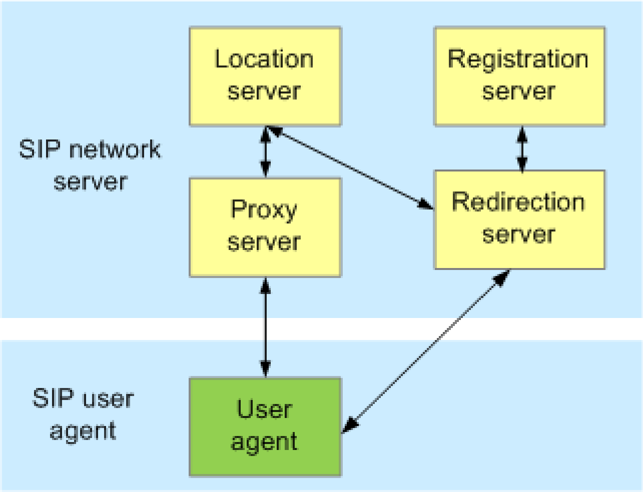
This section describes the SIP protocol that involves network entities, SIP URI, SIP messages, and SIP media negotiation mechanism. Learning of this chapter enables you to have a deep understanding of the SIP protocol.

1.5.2.1 SIP Network Entities and Application

SIP Network Entities

Various logical entities exist on the SIP network to play different roles. Major SIP entities are SIP user agent and SIP network server, as shown in Figure 1-21.

Figure 1-21 SIP network entities



User agent (UA)

The UA sends or receives SIP requests and processes these requests. The UA is logically classified into user agent client (UAC) and user agent server (UAS). The UAC sends requests to UAS, and the UAS responds to these requests. The responses can be acceptance, rejection, or redirection.

NOTE

A Huawei AG functions as a UA.

Proxy server

The proxy server, a logical network entity, forwards requests or responses on behalf of a client. The proxy server can also function as a server. The proxy server provides the following functions: routing, authentication, accounting control, call control, and service providing. The proxy server attempts to forward requests to multiple addresses in multiple modes, such as branching, cycling, and recursively querying.

Registration server

The registration server receives registration requests and saves address mapping contained in the registration requests to the database for the usage by subsequent call processing and subscriber's home address locating.

Redirection server

The redirection server responds to received requests with one or multiple new addresses. Then, the client simply sends requests to these addresses. The redirection server does not receive or reject calls. It mainly completes the routing function and can support the mobility of SIP terminals together with the registration process.

Location server

The location server provides locating functions. The location server obtains the possible called party's address for the redirection server and proxy server and provides a list of mapping between recorded addresses and contact addresses.

In the actual establishment of a SIP application system, a SIP server must cooperate with other background applications to provide a manageable carrier network. For example, the SIP server needs to communicate with the Remote Authentication Dial In User Service (RADIUS) server for terminal authentication.

NOTE

The IMS network or softswitch functions as the redirection server, proxy server, and registration server, and the DNS server functions as the location server.

Basic SIP Functions

SIP provides the following basic functions:

- User location: determines terminals used for communication.
- User capabilities: determines the communication media and parameters used by the media.
- User availability: determines the willingness of the called party to join in the communication.
- Call setup: establishes a call between calling and called parties.
- Call handling: transfers or terminates calls.

SIP AG on the IMS Network

Figure 1-22 shows the position of the SIP AG on the IMS network. NEs that have close relationship with the AG include call session control function (CSCF), proxy-call session control function (P-CSCF), interrogating-call session control function (I-CSCF), serving-call session control function (S-CSCF), home subscriber server (HSS), subscription locator function (SLF), media resource server (MRS), and application server (AS).

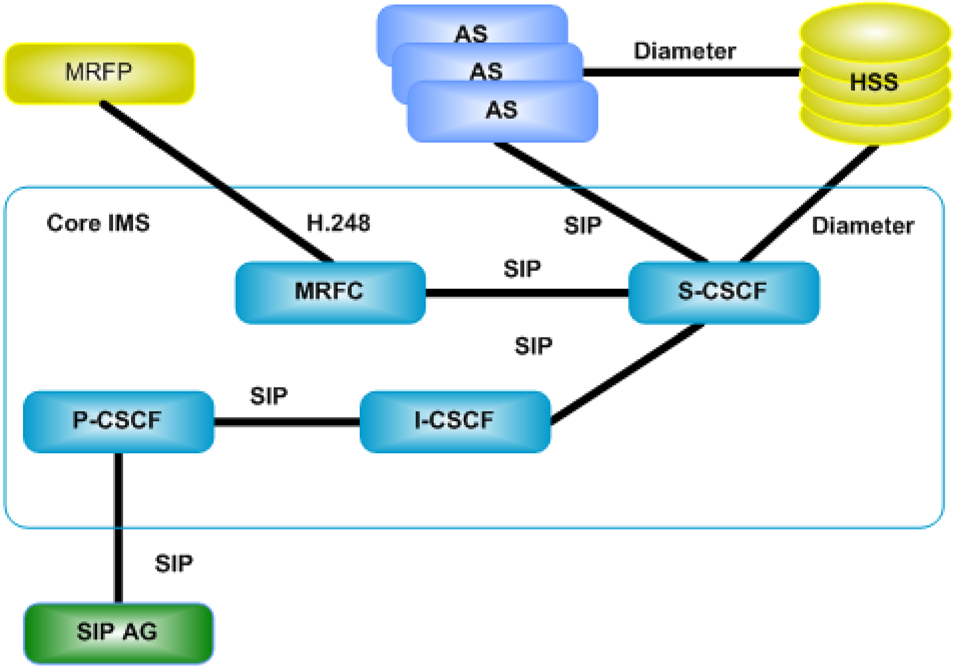
- CSCF: The call control center of the IMS system. The CSCF dispatches multiple real-time services on the IP transmission platform and provides central routing engine, policy management, and policy implementation.
- P-CSCF: The initial contact point between subscribers and the IMS. The P-CSCF routes terminal requests to a correct I-CSCF or S-CSCF and generates CDRs for roaming subscribers. It provides SIP compression on the Gm interface and integrity protection.
- I-CSCF: During the IMS terminal registration, the I-CSCF assigns an S-CSCF for processing subscriber services and locates the S-CSCF that the called party registers with.
- S-CSCF: The S-CSCF provides registration, authentication, service triggering and control, and session routing functions for IMS subscribers.
- HSS: The HSS functions as a database to store information, such as subscriber numbers.

- SLF: When multiple HSS devices exist in the domain, the SLF selects an HSS for storing subscriber data.
- MRS: It is classified into multimedia resource function controller (MRFC) and multimedia resource function processor (MRFP). The MRS plays tones and announcements, processes conference media stream (audio mixing) and DTMF digits, and converts codecs. In some situations, the MRS and AS functions can be played by one device.
- AS: It triggers services.

Figure 1-22 shows voice service providing using the SIP protocol on the IMS network.

- The SIP AG accesses the IMS network through the P-CSCF. The P-CSCF forwards SIP messages (including registration, multimedia session, and IM/Presence messages) to the homed S-CSCF (based on registration information) or I-CSCF (based on the home domain name carried in the SIP message).
- The I-CSCF assigns an S-CSCF for processing subscriber information based on subscriber registration information and CSCF capability information.
- The S-CSCF receives registration requests sent by IMS subscribers and forwarded by the P-CSCF, cooperates with the HSS to authenticate subscribers, and provides routing functions for calling and called parties. The IMS communicates with the AS and MRFC under the control of the S-CSCF.

Figure 1-22 Simplified IMS networking



1.5.2.2 SIP URI

The SIP protocol uses the uniform resource identifier (URI) to identify terminal users. RFC 2369 defines URI rules and syntax.

SIP URI includes SIP URI and TEL URI, either of which can uniquely identify a SIP user. The SIP URI configured on the access device and the IMS for a SIP user must be the same.

SIP URI

SIP URI is used in SIP messages, indicating the initiator of a request (From), the current destination address (Request-URI), the final receiver (To), and the address after redirection (Contact). SIP URI can also be embedded into the Web page or other hyper links to indicate that a certain user or service can be accessed through SIP.

Generally, the SIP URI is in the following format: sip:user:password@host:port;URI-parameters?headers

Table 1-7 SIP URI format description

Item	Description
SIP	Indicates that the SIP protocol is used to communicate with the peer end.
user	Indicates the user name, which can consist of any characters. In general, the user name can be an e-mail address or a telephone number.
password	Indicates the password, which can be presented in the SIP URI. However, password presentation in the SIP URI is not recommended, because it poses security risks.
host	Indicates the host name. It can be the host domain name or an IPv4 address.
port	Indicates the ID of the port to which requests are sent. The default value is **5060**, which is the ID of the public SIP port.
URI-parameters	Indicates URI parameters. • transport-param: specifies the protocol used in the transmission layer, such as, UDP or TCP. • user-param: identifies whether the user name is a telephone number or a common user name. • method-param: specifies the method that is used. • ttl-param: indicates the time-to-live (TTL) value of UDP multicast packets. The parameter is used only when the transmission protocol is UDP and the server address is a multicast address. • maddr-param: indicates the address of the server that communicates with the user. The parameter overwrites any address derived from the host field. Generally, the parameter is a multicast address. **NOTE** Parameters transport-param, method-param, ttl-param, and maddr-param are all URL parameters. They are used only in a redirected address, that is, the Contact header field.
headers	Is contained in requests. The header field can be specified using a question mark (?) in a SIP request. For example, sip:alice@example.huawei.com?priority=urgent

Table 1-8 shows examples of the SIP URI.

Table 1-8 SIP URI examples

Example	Description
sip:55500200@10.10.10.1;	**55500200** indicates the user name, and **10.10.10.1** indicates the IP address of the gateway for IP calls.
sip:55500200@ 10.10.10.1 :5061; User=phone;	**55500200** indicates the user name, **10.10.10.1** indicates the IP address of the host, and **5061** indicates the port ID of the host. **User=phone** indicates that the user name is a telephone number.
sips:1234@10.110.25.239	**sips** indicates the secure SIP URI, that is, the security-based TLS protocol is used on the transmission layer. **1234** indicates the user name, and **10.110.25.239** indicates the IP address of the gateway for IP calls.

TEL URI

TEL URI identifies a telephone number that occupies resources. The telephone number can be a global number or a local number. The global number must comply with the E164 coding standard and start with a plus sign (+). The local number must comply with local private numbering plan. Format:

tel:+86-755-6544487

tel:45687;phonecontext=example.com

tel:45687;phonecontext=+86-755-65

1.5.2.3 SIP Message

Format

The SIP message is encoded in the text format, each line ending with CR or LF. The SIP message has two types, the request message and the response message. The general message format is as follows:

SIP message = Start line
*Message header field
Empty line (CRLF)
[Message body]

A SIP message consists of a start-line, one or more header fields, and a message body. The request and response messages are the same in the format and only differ in the start-line. The request message has a request-line as the start-line and the response message has a status-line as the start-line.

Request Message

Request messages are sent from the client to the server. SIP request messages include INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, PRACK, and UPDATE. Table 1-9 describes these SIP request messages.

Table 1-9 SIP request messages

Request Message	Function
INVITE	Invites a user to join a call
ACK	Acknowledges the response message of the request
OPTIONS	Requests for querying capability information
BYE	Releases an established call
CANCEL	Releases an unestablished call
REGISTER	Registers the user location information on the SIP network server
PRACK	Acknowledges a reliable provisional response message
UPDATE	Updates the session

start-line

The start-line consists of Method, Request-URI, and SIP-Version.

- Method: determines the type and purpose of a request message. Keywords are request messages in Table 1-9.
- Request-URI: identifies the user or server address used by the request.
- SIP Version: indicates the SIP version contained in a request or response. This parameter is case insensitive. In general, values are always in upper cases.

Header field

For details, see Message Header.

Message body

For details, see Message Body.

Response Message

The SIP response message is used for responding to the SIP request message, indicating whether the call is successful or fails. Different from request messages, the start-line of response messages is also called status-line, which consists of SIP-Version, Status-Code, and Reason-Phrase.

- SIP-Version: indicates the used SIP version.
- Status-Code: identifies the response message type. The status-code is a 3-digit integer. The first digit of the status-code defines the response type. The other two digits provide detailed descriptions about the response. Table 1-10 describes response messages.

- Reason-Phrase: provides descriptions about the status code. This field is optional.

Table 1-10 SIP response messages

Status Code	Meaning	Function
1XX	Provisional	The request has been received and is being processed.
2XX	Success	The action was successfully received, understood, and accepted.
3XX	Redirection	Further action needs to be taken in order to complete the request.
4XX	Client Error	The request contains bad syntax or cannot be fulfilled at this server.
5XX	Server Error	The server failed to fulfill an apparently valid request
6XX	Global Failure	The request cannot be fulfilled at any server.

NOTE

Except 1XX responses, other responses are final responses and can terminate requests.

SIP requires that the application must understand the first integer of the response status code, and allows the application not to process the last two integers of the status code.

For example, SIP/2.0 200 OK

- SIP/2.0 indicates that the SIP version is 2.0.
- 200 is the status code, indicating a successful response.
- OK is the cause code and is the further explanation about 200.

Message Header

The SIP message header consists of SIP header fields to complete information transfer and parameter negotiation for SIP sessions. RFC 3261 defined various SIP header fields. This section only introduces five mandatory header fields in a SIP message.

Table 1-11 Mandatory SIP header fields

Header Field	Function	Common Format	Description
Call-ID	Globally identifies a session.	Local flag@host	-
From	Indicates the initiator of the request. The server copies this field from the request message to the response message.	Displayed name<SIP-URI>;tag=XXXX	The tag, a hexadecimal character string, is used to identify two subscribers who share a SIP address to initiate

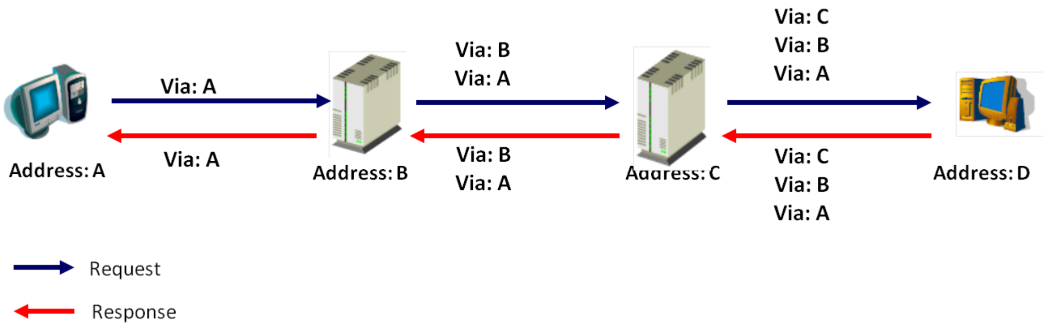
Header Field	Function	Common Format	Description
To	Indicates the recipient of the request. It has the same format as From. To and From differs only in the first keyword.	Displayed name<SIP-URI>;tag=XXXX	calls using the same call ID. The tag value must be unique globally. A subscriber must have the same call ID and tag value for a call.
CSeq	Indicates the sequence number of a request. The client adds this field to each request. The server copies the CSeq value in the request to the response. CSeq is used to determine mapping between responses and requests.	sequence number message name	-
Via	Indicates the path that the request passes. This header field can prevent loop from happening during the request transmission and ensure that the response and request are transmitted in the same path to meet specified requirements.	transmit protocol, sender; hidden parameter, TTL parameter, multicast address parameter, receiver flag, branch parameter	For details, see Via processing.

Via processing

Figure 1-23 shows the Via processing.

- For a request: When transmitting a request, the SIP entity adds its address on the most outer of the Via set of the request. Therefore, when the request reaches the destination, a Via header field set similar to the stack is formed in the request.
- For a response: The destination entity copies the Via address in the request to the response. When receiving the response, the proxy server checks whether the Via at the most outer of the Via set is the proxy server's address. If yes, the proxy server deletes the Via, checks the next Via address, and sends the response to the next Via address. If the next Via address does not exist, this response is terminated on this proxy server.

Figure 1-23 Processing of the Via header field



Message Body

The message body is used to negotiate information and parameters during the call establishment. In addition, the message body also transfers authentication information. In general, SIP messages are in Session Description Protocol (SDP) format.

1.5.2.4 SIP Media Negotiation Mechanism

Session Initiation Protocol (SIP) cooperates with the Session Description Protocol (SDP) to complete media negotiation. Negotiated media includes the IP address, port ID, codec, and media channel parameters.

SDP

SDP, a text-based control protocol at the application layer, is used for negotiating media, including media type and codec solution during session establishment. For the SDP message format, see RFC 2327. Descriptions of common SDP lines are as follows.

SDP lines	Description
v line	Indicates the SDP protocol version.
o line	Provides the session initiator (user name and host address), session flag, and session version number.
s line	Indicates the session name. Each session has a unique session name in the session description.
c line	Indicates the linked address. In general, the linked address is the IP address of an AG or a SIP terminal controlled by the IP multimedia subsystem (IMS).
t line	Indicates the start and stop times for a session. The t line is used if a session is active at multiple irregularly spaced times.
m line	Indicates media description. A session may not have media description or have multiple media descriptions. Media can be audio, video, application (such as whiteboard information), data, and control.
a line	Indicates media attribute. A session may not have media attribute or have multiple media attributes. The media attribute can be:

SDP lines	Description
	- media direction: sendonly, recvonly, inactive, or sendrecv - ptme (packetization time) - bandwidth - codec format

Media Negotiation Process

The SIP protocol implements media negotiation based on a simple offer/answer model. In this model, the call initiator informs the call receiver of all supported media formats. The call receiver selects one or multiple media formats to respond the call initiator. Then, media streams are transmitted using negotiated media formats.

Through the negotiation, both the call initiator and call receiver obtain peer media attributes so that they can communicate using correct media channels.

Offer/answer processes cannot be overlapped. Each media negotiation must be on the basis of the previous negotiation. Media streams can be added to (not deleted from) a new SDP offer. Nevertheless, you can set the port number of the media streams to 0 to change the number of media streams that are used.

Media Negotiation Example

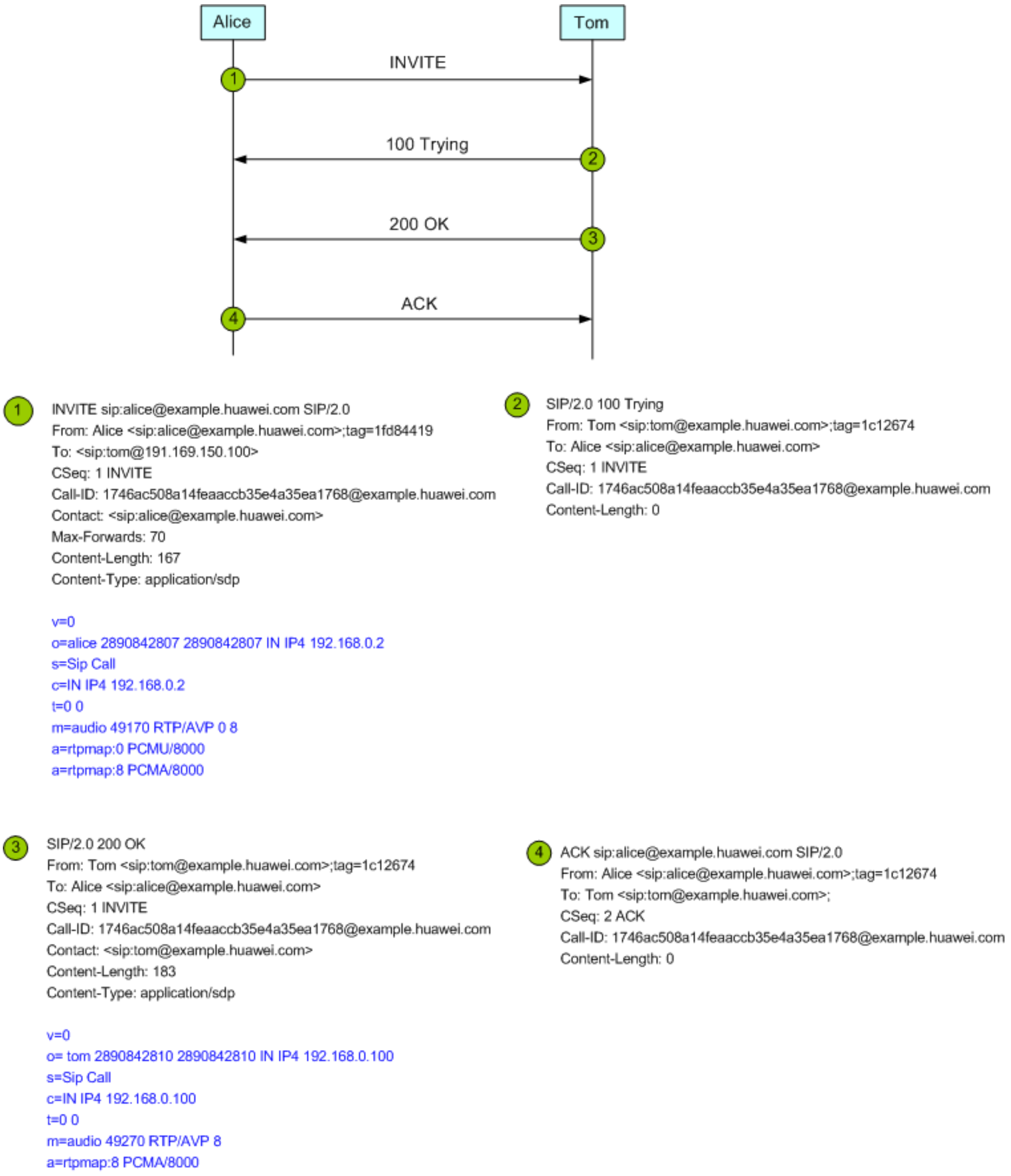
Figure 1-24 shows a SIP media negotiation example.

1. The Alice's SIP phone sends an INVITE message to the Tom's SIP phone to establish a session. The INVITE message contains Alice's SDP information (in blue). The SDP information contains IP address 192.168.0.2, port number 4917, and supported codecs PCMU and PCMA.
2. Tom's SIP phone replies with a 100 response, indicating that the invitation is being processed.
3. Tom's SIP phone sends a 200 response to Alice's SIP phone. The 200 response indicates that the invitation has been successfully processed and carries Tom's SDP information (in blue). The SDP information contains IP address 192.168.0.100, port ID 49270, and supported codec PCMA.
4. After receiving the 200 response, Alice's SIP phone sends an ACK message to Tom's SIP phone to confirm the receiving of the 200 response. Then, Alice and Tom can exchange RTP media packets using the negotiated IP address, port, and codec to communicate with each other.

NOTE

After the session is established, if the Tom's or Alice's SIP phone wants to modify parameters, they can send a Re-INVITE message to initiate another negotiation. The processing rule is the same as that for processing the INVITE message.

Figure 1-24 SIP media negotiation example



1

```
INVITE sip:alice@example.huawei.com SIP/2.0
From: Alice <sip:alice@example.huawei.com>;tag=1fd84419
To: <sip:tom@191.169.150.100>
CSeq: 1 INVITE
Call-ID: 1746ac508a14feaaccb35e4a35ea1768@example.huawei.com
Contact: <sip:alice@example.huawei.com>
Max-Forwards: 70
Content-Length: 167
Content-Type: application/sdp

v=0
o=alice 2890842807 2890842807 IN IP4 192.168.0.2
s=Sip Call
c=IN IP4 192.168.0.2
t=0 0
m=audio 49170 RTP/AVP 0 8
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
```

2

```
SIP/2.0 100 Trying
From: Tom <sip:tom@example.huawei.com>;tag=1c12674
To: Alice <sip:alice@example.huawei.com>
CSeq: 1 INVITE
Call-ID: 1746ac508a14feaaccb35e4a35ea1768@example.huawei.com
Content-Length: 0
```

3

```
SIP/2.0 200 OK
From: Tom <sip:tom@example.huawei.com>;tag=1c12674
To: Alice <sip:alice@example.huawei.com>
CSeq: 1 INVITE
Call-ID: 1746ac508a14feaaccb35e4a35ea1768@example.huawei.com
Contact: <sip:tom@example.huawei.com>
Content-Length: 183
Content-Type: application/sdp

v=0
o= tom 2890842810 2890842810 IN IP4 192.168.0.100
s=Sip Call
c=IN IP4 192.168.0.100
t=0 0
m=audio 49270 RTP/AVP 8
a=rtpmap:8 PCMA/8000
```

4

```
ACK sip:alice@example.huawei.com SIP/2.0
From: Alice <sip:alice@example.huawei.com>;tag=1c12674
To: Tom <sip:tom@example.huawei.com>;
CSeq: 2 ACK
Call-ID: 1746ac508a14feaaccb35e4a35ea1768@example.huawei.com
Content-Length: 0
```

1.5.3 SIP Services and Basic Service Flows

This chapter describes the subscriber registration and authentication flow, subscription flow, and basic call flow.

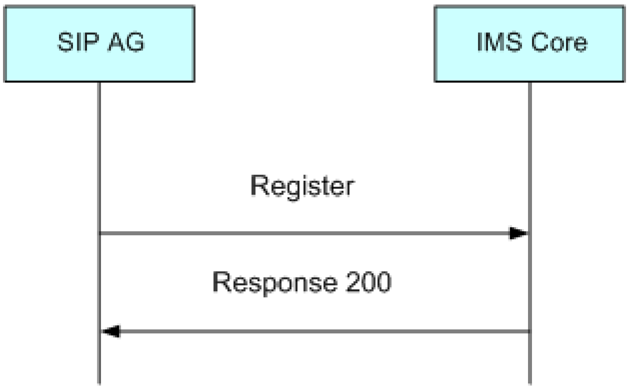
1.5.3.1 User Registration and Authentication Flows

Before initiating a call, a SIP user must register with the home network to map the domain name to an IP address. The registration is of two types: the registration not requiring authentication and the registration requiring authentication. After the system is powered on or after the user is added, the user registration flow is started.

Registration Not Requiring Authentication

As shown in Figure 1-25, the SIP AG sends the REGISTER request message to the IMS for each user. The message contains information such as the user ID. After receiving the REGISTER request message, the IMS checks whether the user is already configured on the IMS. If the user is already configured, the IMS responds to the SIP AG with the RESPONSE 200 message.

Figure 1-25 Flowchart of the registration not requiring authentication

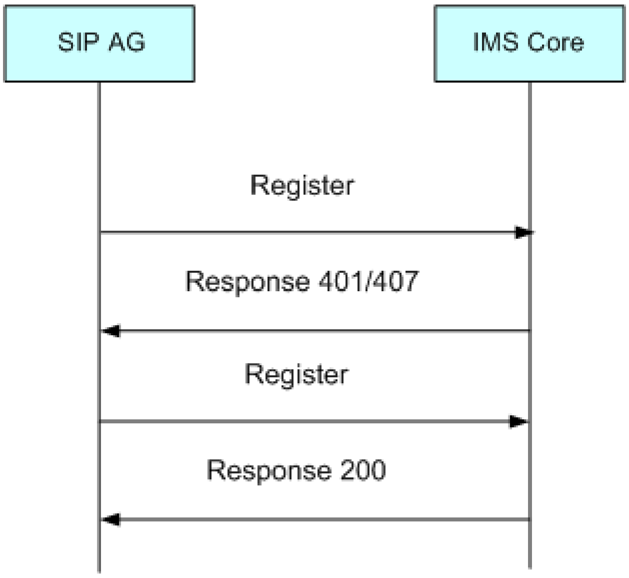


Registration Requiring Authentication

As shown in Figure 1-26, the SIP AG sends the REGISTER request message to the IMS for each user. The message contains information such as the user ID.

The IMS responds with the RESPONSE 401/407 message, the message containing information such as the key and the encryption mode. The SIP AG encrypts the corresponding user name and password, generates a new REGISTER request message, and sends the message to the IMS. The IMS decrypts the message and verifies the user name and password. If the user name and password are correct, the IMS responds to the SIP AG with the RESPONSE 200 message.

Figure 1-26 Flowchart of the registration requiring authentication



Registration Modes

Table 1-12 shows the registration modes supported by SIP AGs.

Table 1-12 Registration modes supported by SIP AGs

Registration Mode	Description
Separate account registration	A non-wildcard number is used for registration. Each account is registered separately.
Wildcard number registration	A wildcard number, such as 2878*, is used for registration. After the registration, all numbers with prefix "2878" are successfully registered on the IMS.
Proxy group registration	A batch of accounts are added to a group. Then, an account in the group or a separate group account is used for registration. After the registration, all accounts in the group are successfully registered on the IMS.

NOTE

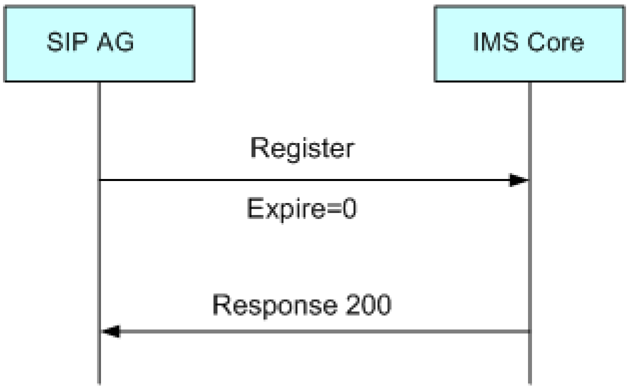
If a large number of accounts concurrently register with the IMS in separate account registration, mass registration messages degrade IMS performance. The wildcard registration and proxy group registration reduce the number of registration messages to be exchanged between AGs and the IMS.

Deregistration

When a SIP AG attempts to deregister from the IMS, the AG sends a REGISTER request with timeout duration set to 0s to the IMS. The deregistration request can also be initiated by the IMS.

Deregistration Initiated from the AG

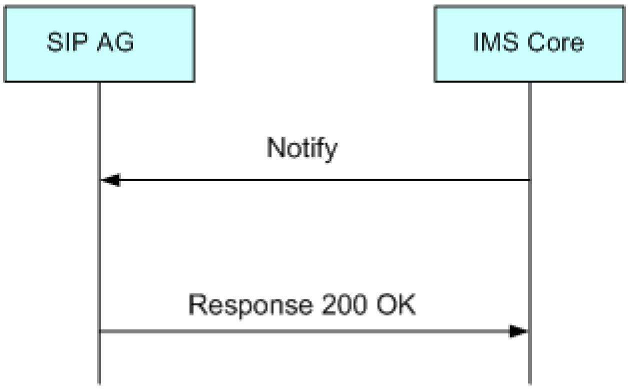
Figure 1-27 Deregistration initiated from the AG



As shown in Figure 1-27, the SIP AG initiates deregistration by sending a new REGISTER request with **Expires** set to 0 to the IMS.

Deregistration Initiated from the IMS

Figure 1-28 Deregistration initiated from the IMS



As shown in Figure 1-28, when the IMS requires to clear the registration of a SIP AG, the IMS sends a NOTIFY message carrying the deregistration reason to the AG if the IMS has subscribed to the registration status of this AG. After receiving the NOTIFY message, the SIP AG responds to the IMS with a 200 OK message.

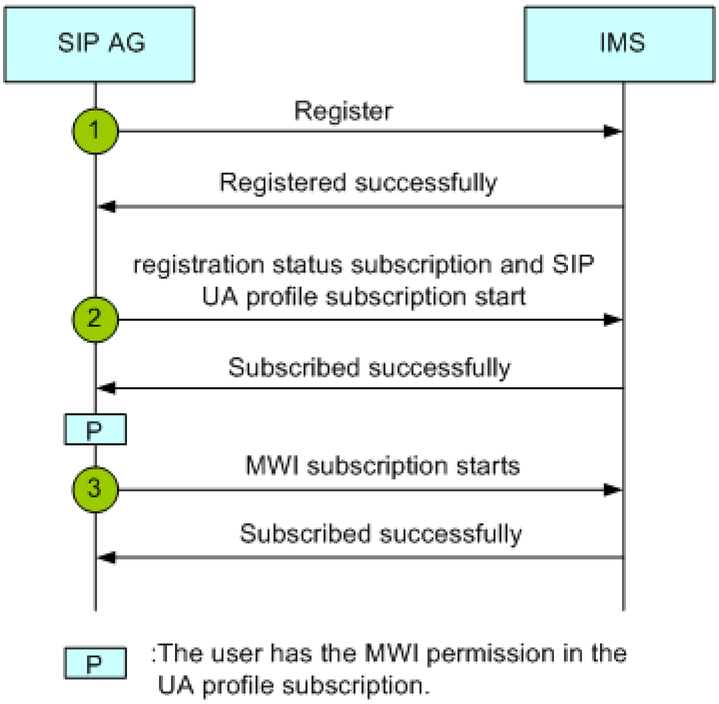
1.5.3.2 Subscription

Subscription indicates that a SIP access gateway (AG) requests the current status, information change, and service permission of a subscription user from the network side. The SIP AG supports the following subscriptions:

- Registration status subscription
- User agent (UA) profile subscription
- Message waiting indication (MWI) subscription

As shown in Figure 1-29, after a user is successfully registered, registration status subscription and SIP UA profile subscription start. If the user has the MWI permission in the UA profile subscription, an MWI subscription starts.

Figure 1-29 Subscription flow

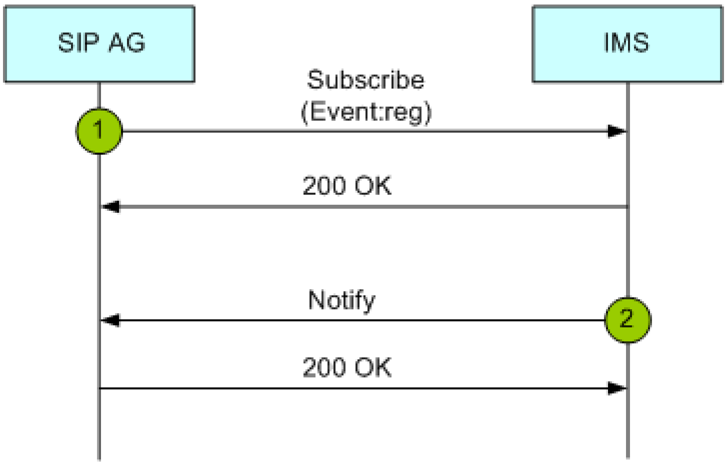

SIP AG
IMS
1
Register
Registered successfully
registration status subscription and SIP UA profile subscription start
2
Subscribed successfully
P
MWI subscription starts
3
Subscribed successfully
P
:The user has the MWI permission in the UA profile subscription.

Registration Status Subscription

Registration status subscription is mainly used to obtain the user's registration status and registration status change. Figure 1-30 shows the flow of a registration status subscription:

1. The SIP AG uses the **Subscribe** message to initiate a registration status subscription and uses the **Event** header to identify the subscription type.
2. The IMS uses the **Notify** message to issue the user's registration status to the SIP AG.

Figure 1-30 Flow of a registration status subscription



UA Profile Subscription

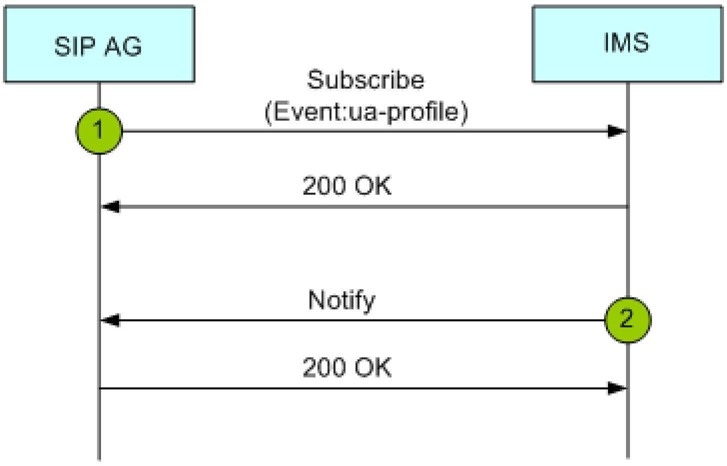
UA profile subscription is used to obtain the user's service permission and dial tone scheme. For details about service permissions supported by a SIP user, see SIP Value-added Services. Figure 1-31 shows the flow of a UA profile subscription.

1. The SIP AG uses the **Subscribe** message to initiate a registration status subscription and uses the **Event** header to identify the subscription type.
2. The IMS uses the **Notify** message to issue the user's registration status to the SIP AG.

NOTE

Users' service permissions can be subscribed to the IMS and can be configured at local through the SIP AG. If the SIP AG is not subscribed to the IMS, service permissions configured on the SIP AG take effect.

Figure 1-31 Flow of a UA profile subscription

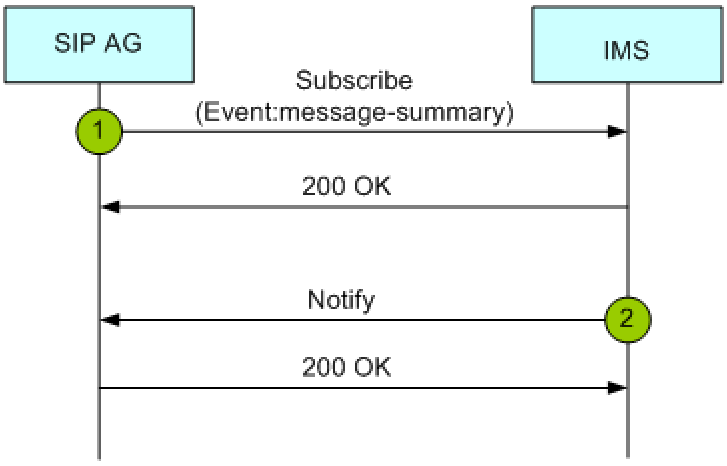


MWI Subscription

MWI subscription is used to obtain the user's messages. Figure 1-32 shows the flow of an MWI subscription.

1. The SIP AG uses the **Subscribe** message to initiate a registration status subscription and uses the **Event** header to identify the subscription type.
2. The IMS uses the **Notify** message to issue the user's registration status to the SIP AG.

Figure 1-32 Flow of an MWI subscription



1.5.3.3 SIP-based VoIP

Figure 1-33 shows the networking for SIP-based voice over IP (VoIP) calls.

Figure 1-33 Networking for SIP-based VoIP calls

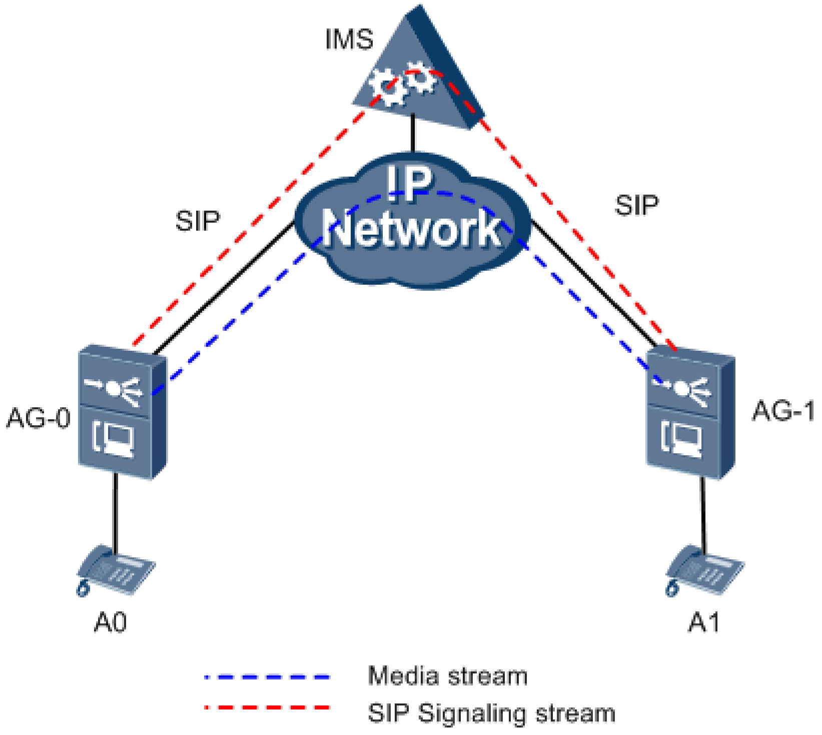
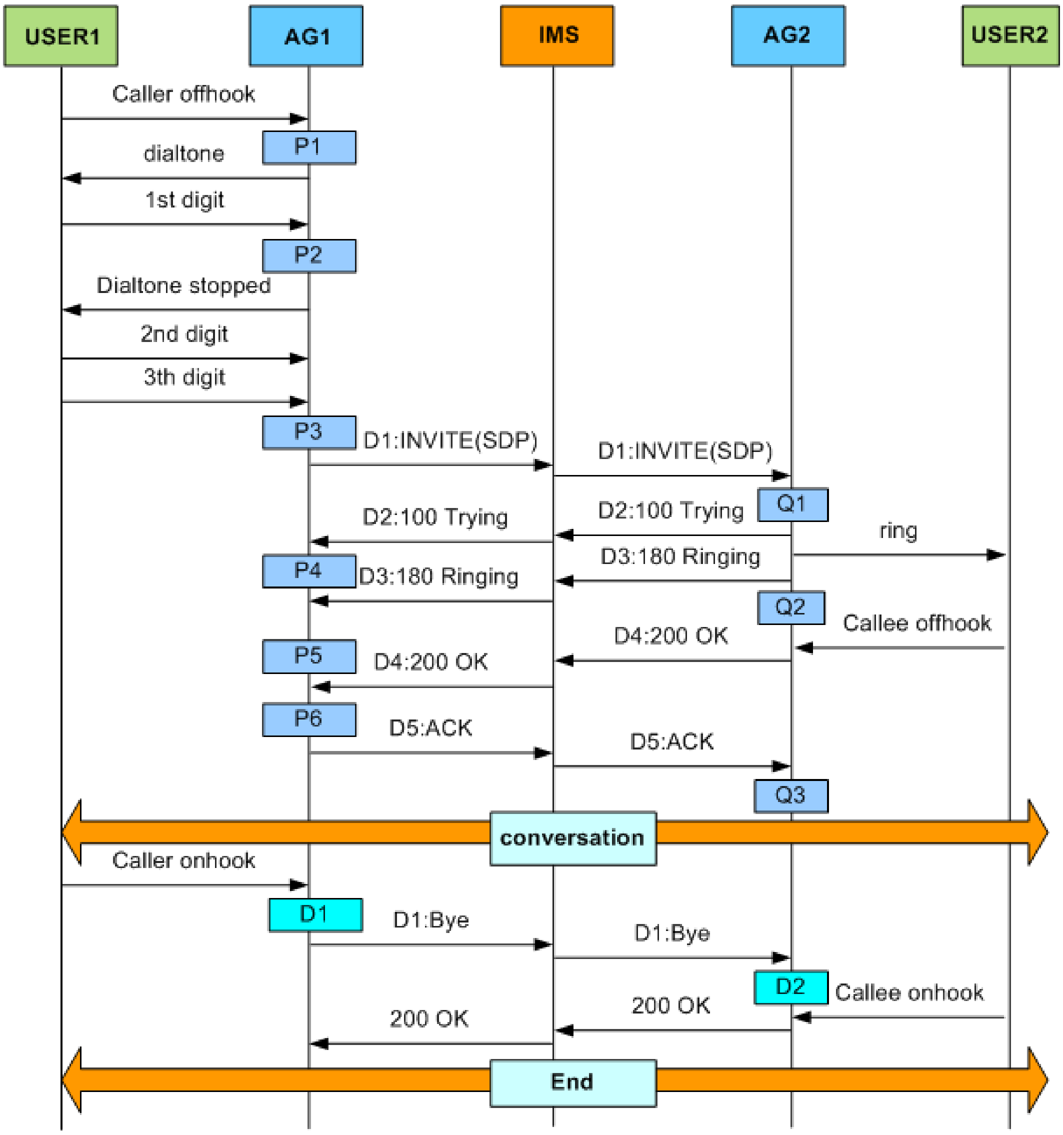


Figure 1-34 shows the SIP-based VoIP call flow. USER1 is the calling party, and USER2 is the called party.

Figure 1-34 SIP-based VoIP call flow



Call flow on the calling side

- P1: AG1 receives the offhook message of USER1 and plays the dial tone to USER1.
- P2: AG1 receives the first dialed digit, stops playing the dial tone, and then starts matching the digit with the digitmaps.
- P3: After receiving N dialed digits and matching the digits with the digitmaps, AG1 finds that the dialed number matches a certain digitmap. Then, AG1 generates the INVITE message and sends the message to the IMS.
- P4: AG1 receives RESPONSE 100 and knows that the peer end receives the INVITE message, so AG1 stops the INVITE message retransmitting flow.
- P5: AG1 receives 180, which indicates that the phone of USER2 is ringing. Then, AG1 plays the ring back tone to USER1.
- P6: AG1 receives 200, which indicates that USER2 answers the phone, so AG1 stops playing the ring back tone to USER1, and changes the stream mode to the bidirectional mode. Then, AG1 constructs an ACK message and sends the message to the IMS.

In the actual processing, when USER1 initiates a call, the IMS determines the situation as follows:

- If USER1 has been configured but not registered with the IMS, the IMS rejects USER1 and responds with 403 to AG1.
- If USER1 is not configured, the IMS rejects USER1 and responds with 404 to AG1.

Call flow on the called side

- Q1: After receiving the INVITE message from the IMS, AG2 replies with a 100 response to the IMS, finds USER2 based on the P-Called-Party-ID, RequestURI, and TO fields (or TEL-URI) contained in the INVITE message, plays the ring tone to USER2, and sends 180 to the IMS, indicating that USER2 is being alerted.
- Q2: After detecting that USER2 picks up the phone, AG2 stops playing the ring tone and sends 200 to the IMS, indicating that USER2 has picked up the phone.
- Q3: After AG2 receives an ACK message, calling and called parties start a session.

In the actual processing, when USER1 initiates a call, AG2 determines the situation as follows:

- If USER2 has been configured but is not registered with the IMS, AG2 replies with 403 to reject the call.
- If USER2 is not configured, AG2 replies with 404 to reject the call.

Call release flow

- D1: After detecting that USER1 hangs up, AG1 sends a BYE message to the IMS and releases DSP resources.
- D2: After receiving the BYE message, AG2 notifies USER2 of the on-hook event. USER2 hangs up, and the call ends.

1.5.3.4 SIP-Based FoIP

In terms of transmission protocol, the fax service can be classified into transparent transmission and T.38; in terms of switching mode, the fax service can be classified into auto-switching and negotiated-switching. Hence, there are four combinations of the fax mode: auto-switching transparent transmission, auto-switching T.38, negotiated-switching transparent transmission, and negotiated-switching T.38.

The working principle of auto-switching is that the AG detects the fax tone, and then selects the transparent transmission or T.38 mode according to the configuration. In this case, the AG needs not send any signaling to the peer device.

The working principle of negotiated-switching is that the AG detects the fax tone, and according to the configuration sends the peer end the re-INVITE message that contains the negotiation parameters for negotiating the fax mode.

In actual application, fax can also be classified into low-speed fax and high-speed fax in terms of transmission speed. The high-speed fax cannot adopt the T.38 mode. A high-speed fax machine can actually be regarded as a modem. With the speed reduced, a high-speed fax machine can also adopt the T.38 mode.

Flow of the Negotiated-Switching Transparent Transmission Fax

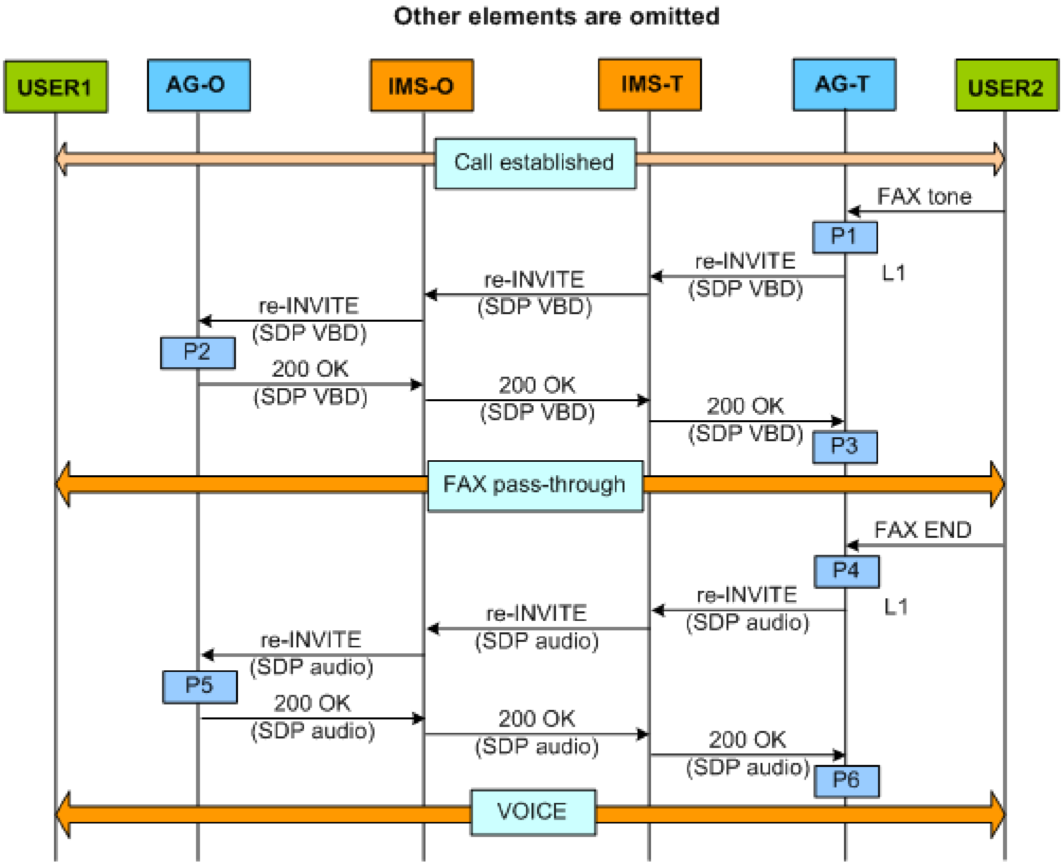
Currently, this fax mode can be presented in three ways.

- Presented as a=fax. This is a G.711 transparent transmission fax mode proposed by China Telecom.
- Presented as a=silenceSupp:off. This is a G.711 transparent transmission fax mode defined in the *draft-IETF-sipping-realtimefax-01.txt*.
- Presented as a=gpmd:99 vbd=yes. This is a VBD mode defined in the ITU-T V.152.

Which method to be applied depends on the parameters configured.

Figure 1-35 shows the fax flow.

Figure 1-35 Flow of the negotiated-switching transparent transmission fax



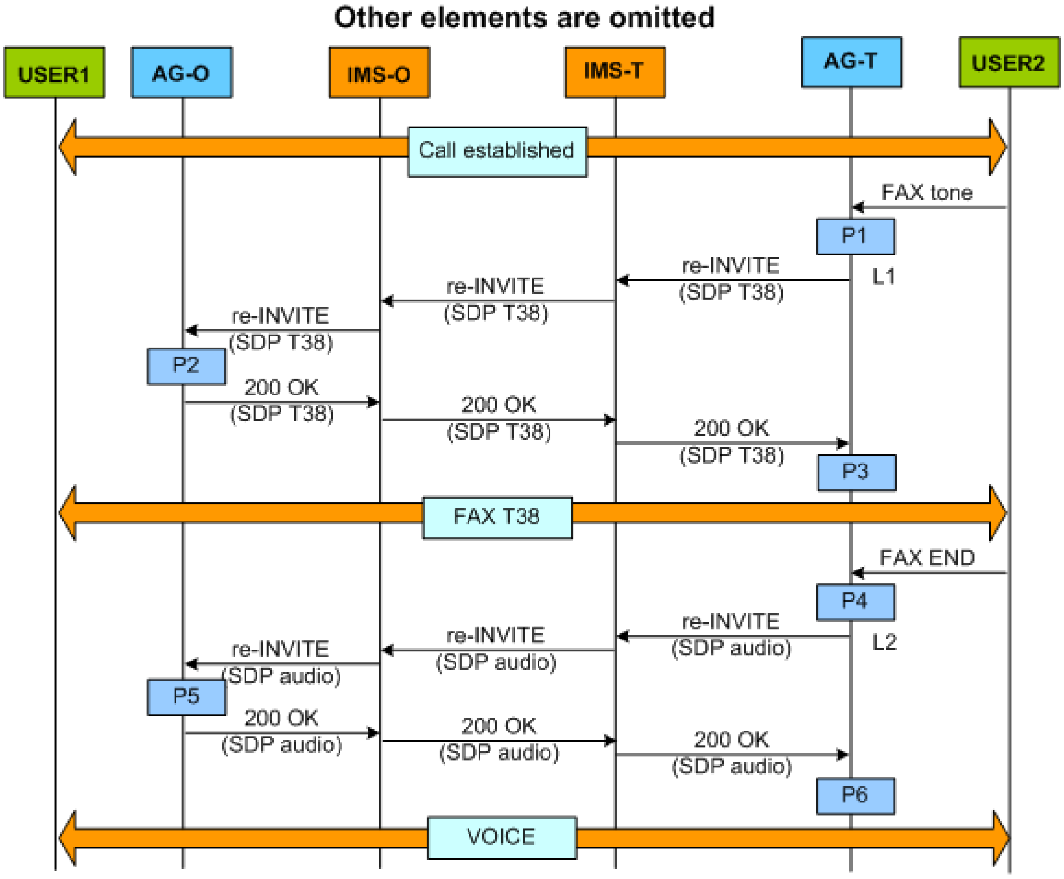
- P1: AG-T first detects the fax tone, and then sends the re-INVITE message to the AG (AG-O) to which the calling party is connected.
- L1: The SDP message contained in the re-INVITE message has three types. The specific fax mode must be configured on the AGs. The initiator of negotiation uses the **a** parameter of different values, and the recipient of negotiation needs to be compatible with the three parameter values. This means that when the recipient receives the re-INVITE message, the recipient should be able to complete the negotiation process with the initiator regardless of the **a** parameter value.
  - The G.711 transparent transmission fax/modem mode defined in the *draft-IETF-sipping-realtimefax-01.txt*.
  - The G.711 transparent transmission fax/modem mode proposed by China Telecom.
  - The VBD mode defined in the ITU-T V.152.

- P2: AG-O receives the re-INVITE message. Then, AG-O generates the 200 OK message and sends the message to AG-T.
- P3: AG-T receives the 200 OK message, and also enables the DSP channel in the fax mode.
- P4: AG-T receives the fax end signal, and sends the re-INVITE message to AG-O.
- L2: The SDP message contained in the re-INVITE message is for setting up a common voice channel.
- P5: AG-O receives the re-INVITE message and switches the DSP channel to the voice mode.
- P6: AG-T receives the 200 OK message, and also switches the DSP channel to the voice mode.

Flow of the Negotiated-Switching T.38 Fax

Figure 1-36 shows the flow of the negotiated-switching T.38 fax.

Figure 1-36 Flow of the negotiated-switching T.38 fax



- P1: AG-T first detects the fax tone, and then sends the re-INVITE message to the AG (AG-O) to which the calling party is connected.
- L1: The SDP message contained in the re-INVITE message carries the T.38 information.

- P2: AG-O receives the re-INVITE message, learns that the peer device requires the T.38 mode, and enables the DSP channel in the T.38 mode. Then, AG-O generates the 200 message and sends the message to AG-T.
- P3: AG-T receives the 200 OK message, and also enables the DSP channel in the T.38 mode.
- P4: AG-T receives the fax end signal, and sends the re-INVITE message to AG-O.
- L2: The SDP message contained in the re-INVITE message is for setting up a common voice channel.
- P5: AG-O receives the re-INVITE message and switches the DSP channel to the voice mode.
- P6: AG-T receives the 200 OK message, and also switches the DSP channel to the voice mode.

NOTE

Figure 1-37 and Figure 1-38 shows the fax flows when the peer device does not support the T.38 mode.

Figure 1-37 Flow of the negotiated-switching T.38 fax when the peer device does not support the T.38 mode (scenario 1)

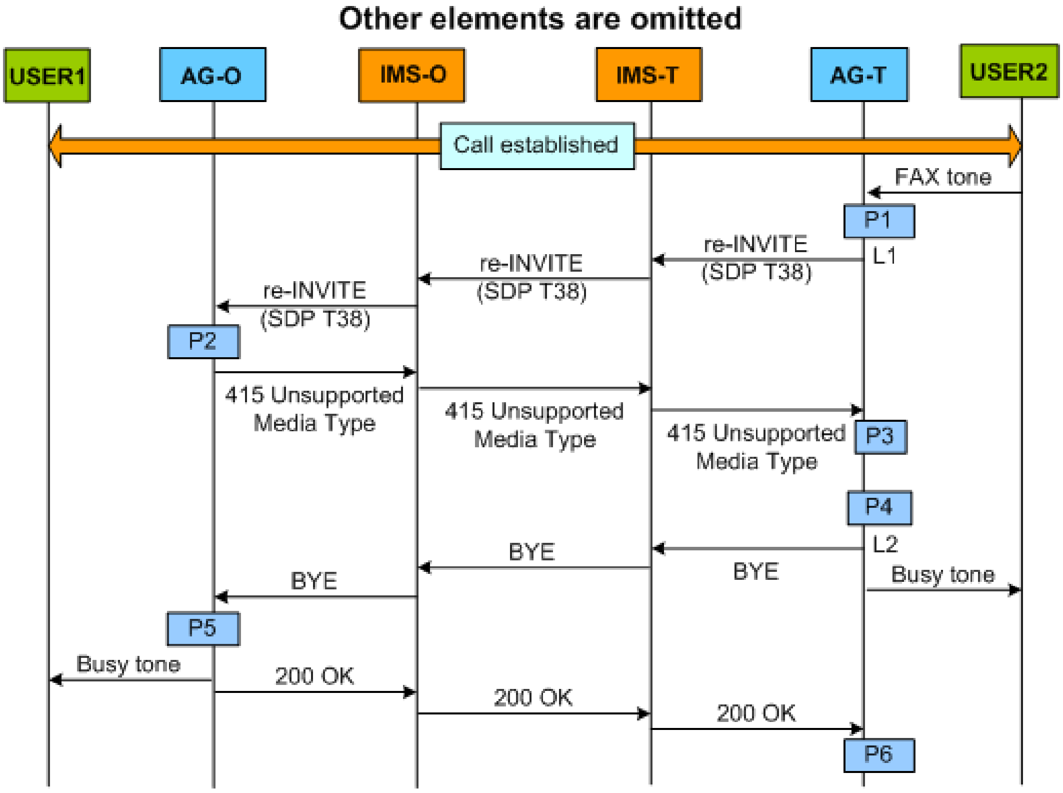
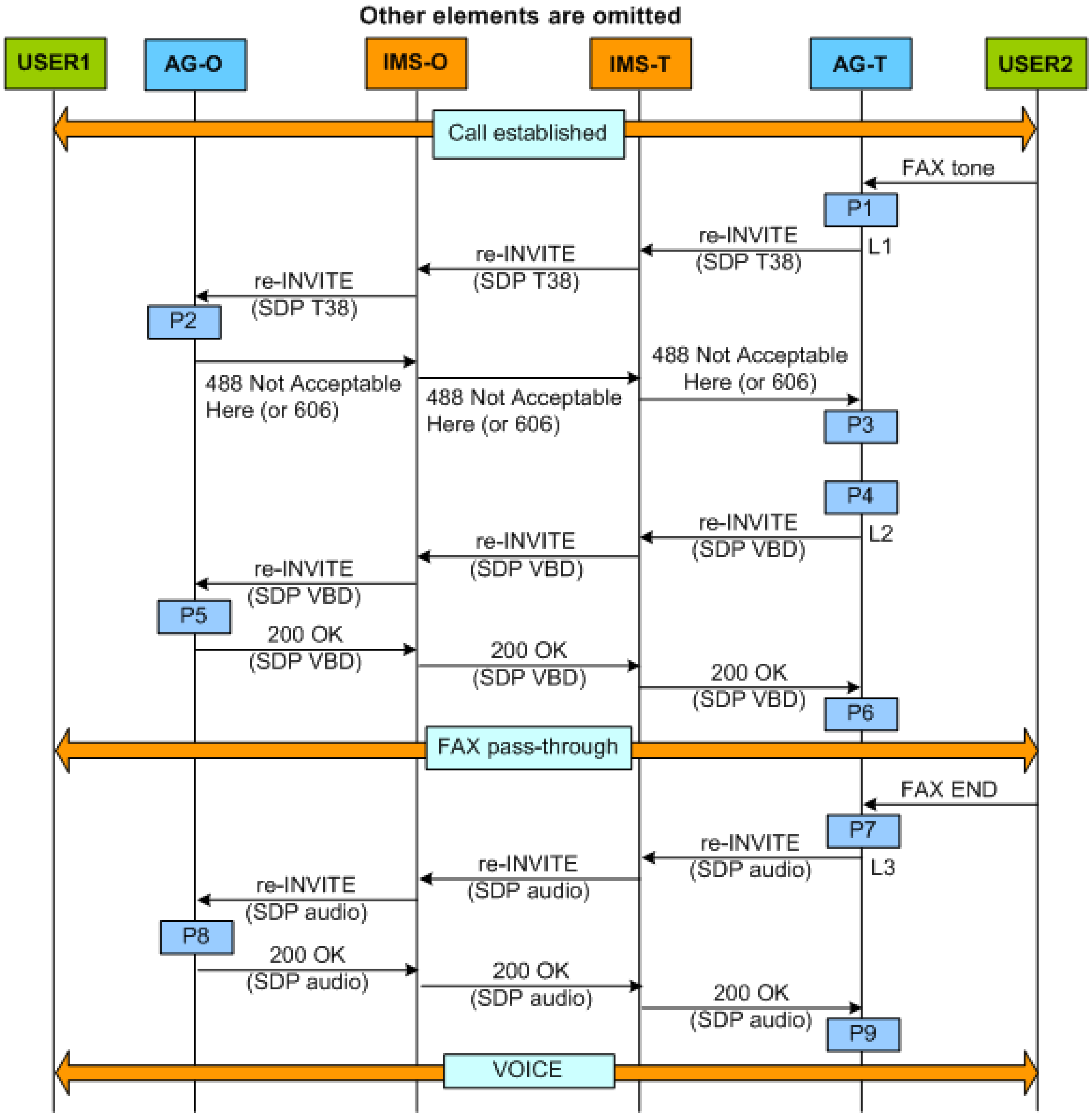


Figure 1-38 Flow of the negotiated-switching T.38 fax when the peer device does not support the T.38 mode (scenario 2)



In scenario 1, if AG-O does not support T.38, it may respond with 415 Unsupported Media Type. After AG-T receives the 415 response, AG-T sends the BYE message and releases the current call. In scenario 2, if AG-O does not support T.38, it responds with 488 Not Acceptable Here or 606 Not Acceptable. After AG-T receives the 488/606 response, AG-T generates another re-INVITE message. The SDP message in this message contains the VBD media type. Thus, the negotiation on the T.38 mode fails, and the transparent transmission mode is adopted.

The access device supports the T.38 mode, and therefore does not respond with the 415/488/606 message in the T.38 negotiation. The access device, however, can process such error codes sent by the peer device.

Flow of the Auto-Switching Transparent Transmission Fax

Generally, the called fax terminal detects the fax tone on the TDM side first, and the calling fax terminal detects the fax tone sent from the IP side. The fax terminal that detects the fax tone automatically switches to the transparent transmission mode without the SIP negotiation.

One problem currently exists in the auto-switching fax flow: If the DSP channel originally works in the G.729 mode for the voice service, and is now switched to the G.711 transparent transmission mode when the fax tone is detected, the G.711 voice packet may not be recognized. This is because the DSP channel of the calling party stills works in the G.729 mode. Therefore, the DSP chip is required to be able to receive G.711 packets when working in the G.729 or other coding modes. The prerequisite remains that the DSP chip should detect and report the fax tone sent from the IP side.

Flow of the Auto-Switching T.38 Fax

The working principle of this fax flow is the same as the working principle of the auto-switching transparent transmission fax. The difference is that, after the fax tone is detected, the DSP channel is enabled in the T.38 mode instead of the transparent transmission mode.

1.5.3.5 SIP-Based MoIP

In terms of service flow, the modem service is similar to the transparent transmission fax service, and can also be classified as auto-switching and negotiated-switching.

The modem service in the negotiated-switching transparent transmission mode can be presented in three ways.

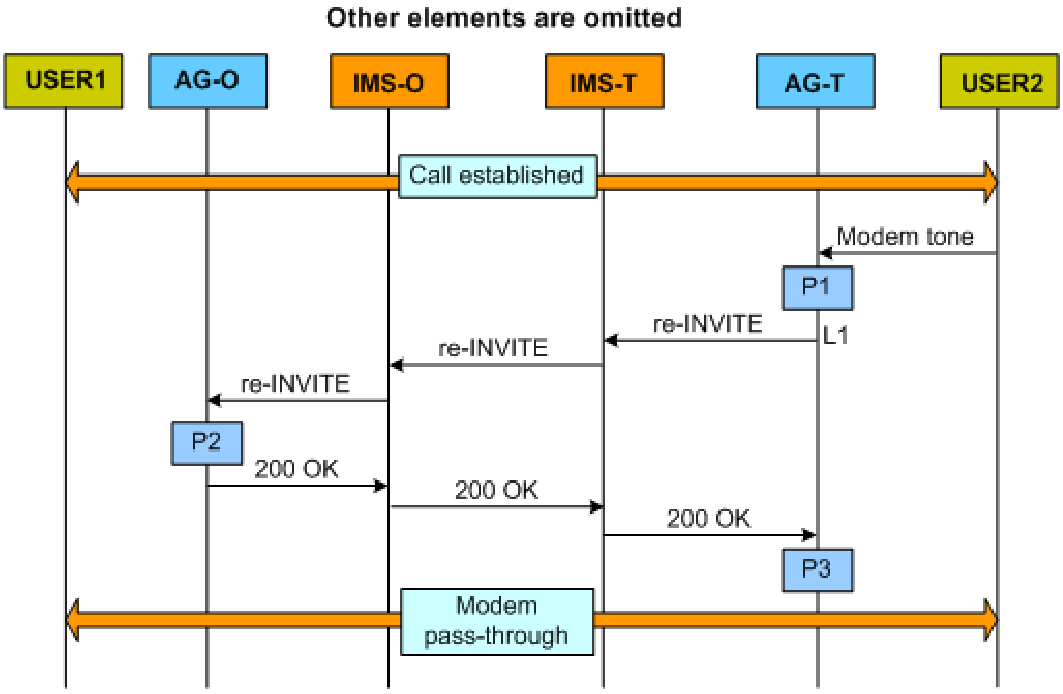
- Presented as a=modem. This is a G.711 transparent transmission modem mode proposed by China Telecom.
- Presented as a=silenceSupp:off. This is a G.711 transparent transmission modem mode defined in the *draft-IETF-sipping-realtimefax-01.txt*.
- Presented as a=gpmd:99 vbd=yes. This is a VBD mode defined in the ITU-T V.152.

The method actually applied depends on the parameters configured.

Flow of the Negotiated-Switching Modem Service

Figure 1-39 shows the flow of the negotiated-switching modem service.

Figure 1-39 Flow of the negotiated-switching modem service



- P1: AG-T first detects the modem tone, and then sends the re-INVITE message to the AG (AG-O) to which the calling party is connected.
- L1: The SDP message contained in the re-INVITE message has three types, corresponding to the three preceding presentations of the negotiated-switching transparent transmission mode. The specific transparent transmission modem mode must be configured on the AGs.
- P2: AG-O receives the re-INVITE message. Then, AG-O generates the 200 message and sends the message to AG-T.
- P3: AG-T receives the 200 OK message, and also enables the DSP channel in the fax or modem mode.

Auto-Switching Modem Mode

In this mode, after the AG detects the modem tone, the AG automatically switches the DSP channel to the VBD mode without notifying the IMS or the peer device.

Generally, the called modem detects the modem tone on the TDM side first, and the calling modem detects the modem tone sent from the IP side. The modem that detects the modem tone automatically switches to the VBD mode without the SIP negotiation.

1.5.3.6 SIP Trunking

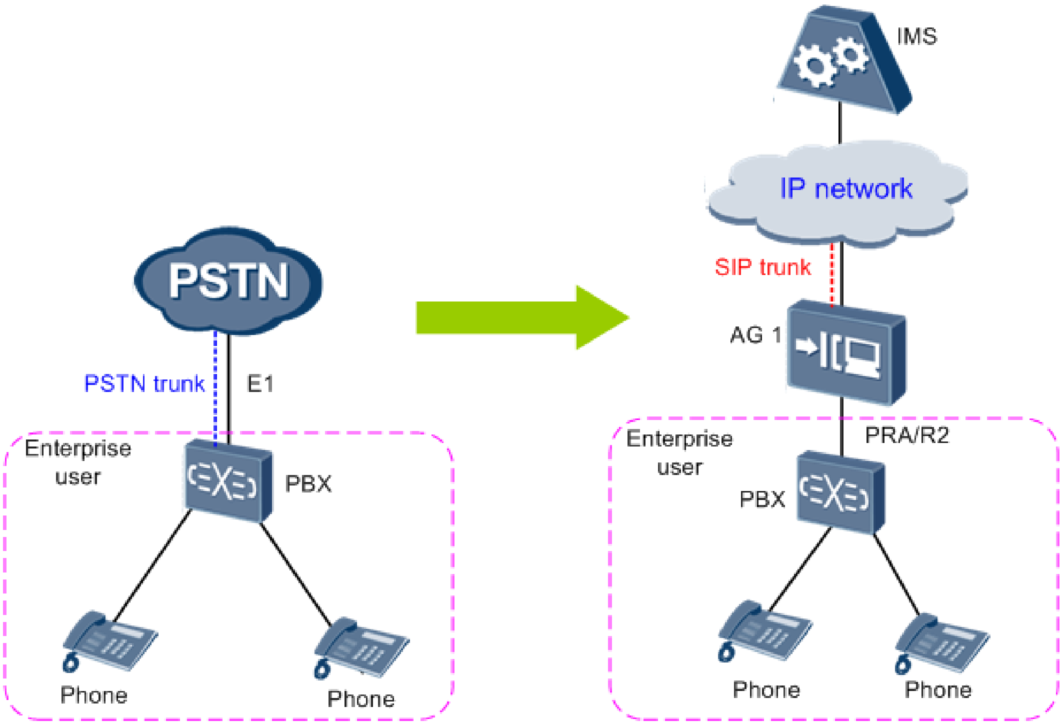
This section describes the SIP trunking feature and its typical usage scenarios.

Background

The traditional private branch exchanges (PBXs) of government, enterprise, and business users connect to carrier networks using PSTN trunk lines (E1 lines). The All-IP network upgrade and reconstruction urgently demands an IP trunk technology to replace the traditional

PSTN trunk technology. Then SIP trunking technology can meet such requirements, connecting government, enterprise, and business PBXs to carrier networks for voice and multimedia services. Figure 1-40 shows the network reconstruction for SIP trunking.

Figure 1-40 Network reconstruction for SIP trunking



Basic Concepts

- Trunk: a physical communication line connecting two switching systems for carrying media stream signals, such as voice, data, and video signals.
- The SIP trunking technology uses SIP to connect government, enterprise, and business PBXs to carrier networks over an IP network.
- SIP trunking allows a PBX to register with the IMS in one of the following modes:
  - Separate number registration: In this mode, the PBX uses a separate number but not a wildcard number to register with the IMS.
  - Wildcard number registration: When the PBX successfully registers with the IMS using a wildcard number, such as 2878*, all numbers with prefix "2878" are successfully registered on the IMS.
  - Agent group registration: In this mode, multiple numbers are added to one group and the PBX uses one of the numbers in this group or a separate group number to register with the IMS. If the registration is successful, all numbers in this group are successfully registered on the IMS. This registration mode reduces registration message exchanging between the IMS and the AG connected to this PBX.
  - No registration: In this mode, the PBX supports call originating without registration. This mode is used if the access network is reliable.
- SIP trunking supports call originating in one of the following modes:
  - Direct dialing in (DDI) calling: In this mode, the PBX requires multiple extension numbers and uses a separate number to register with the IMS; or the PBX uses a

wildcard number to register with the IMS. The extension phones connected to the PBX can call each other using either long or short numbers. External users can call an extension phone by dialing its long number. User experience in DDI calling mode is the same as that in POTS access mode.

- Global dialing number (GDN) calling: In this mode, the PBX requires only a switchboard number, also called "pilot number". The extension phones connected to the PBX call each other using short numbers. When an external user wants to call an extension phone, the user must dial the switchboard number up and the switchboard transfers to the call to the extension phone. In GDN calling mode, the number displayed for the external user is not the number of the extension phone where the call is initiated but the number of the switchboard number. The external user cannot call this extension phone in callback mode.
- Line hunting: In this mode, multiple E1 lines connected to the PBX are added to one hunting group and one number is configured for this group. One hunting group can have one or multiple group numbers. When the PBX registers with the IMS, the AG adds these group numbers to one agent group for registration. For details about line hunting call process, see Line hunting.
- Call routing identified by **tgrp** and **trunk-context**: This mode is used if call routing and charging cannot be implemented using only phone numbers. **tgrp** and **trunk-context** are defined in RFC 4904. They are only used in the Request URI and Contact URI, functioning as **useinfo** in the SIP URI or **par** in the TEL URI. The two parameters must be used in couple. Otherwise, AG considers that the URI does not carry this group of parameters. Huawei AG devices use these two parameters in hunting groups for call routing and charging.

Typical Usage Scenarios

Scenario 1: DDI calling

Usage scenario: As shown in Figure 1-41, the enterprise user uses one PBX to connect to 30 extension phones and the PBX connects to the AG using one PRA port. The 30 extension phones use different phone numbers, 28780001 through 28780030.

Configuration: Wildcard number 2878* is configured for the PRA port and the PBX uses this wildcard number to register with the IMS.

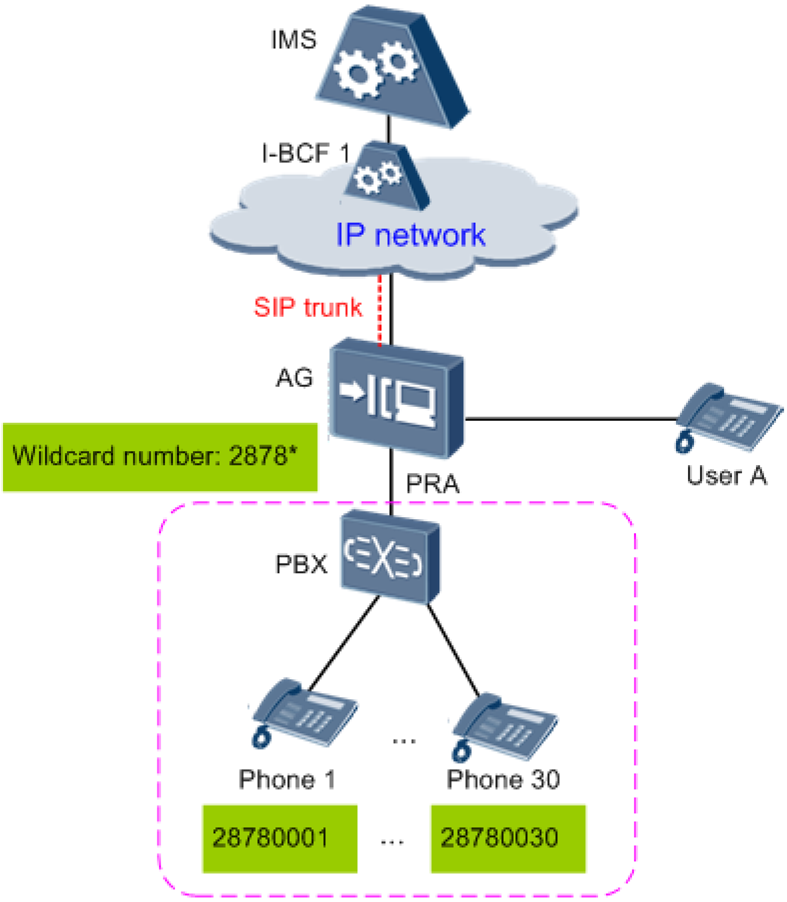
NOTE

If the IMS does not support the registration using a wildcard number, configure 1 primary number and 29 extension numbers for the PBX. All the 30 numbers require a separate number registration.

Call description:

- When user A dials number 28780001 up, phone 1 is called without requiring call transferring from the switchboard.
- When phone 1 calls user A, the number displayed for user A is 28780001, the number of phone 1.
- Phone 1 can call phone 30 using either long number 28780030 or short number 0030 of phone 30.

Figure 1-41 DDI calling



Scenario 2: GDN calling

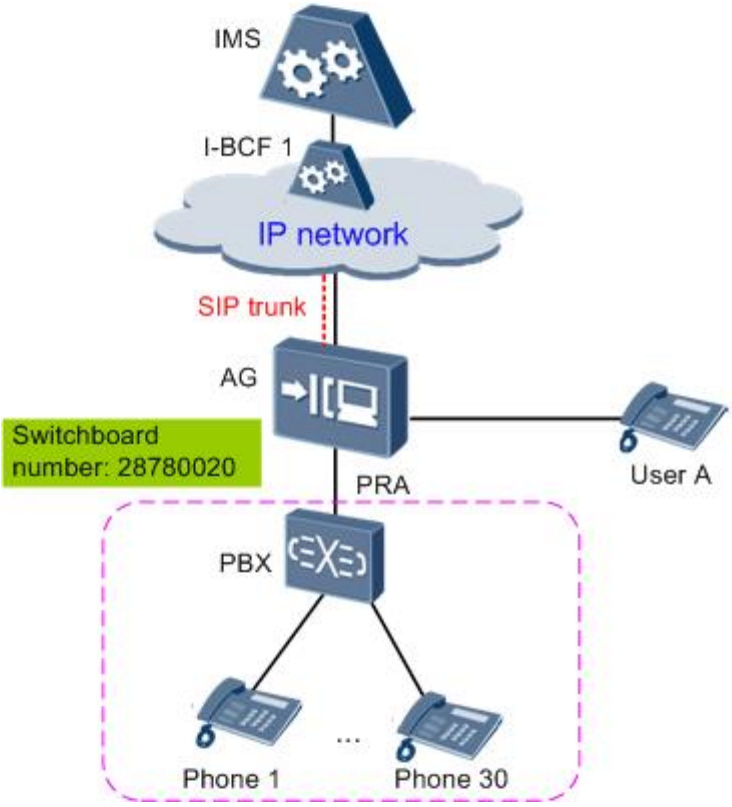
Usage scenario: As shown in Figure 1-42, the enterprise user uses one PBX to connect to 30 extension phones and the PBX connects to the AG using one PRA port. The external number of the 30 extension phones is 28780020.

Configuration: Number 28780020 is configured for the PRA port and the PBX uses this number to register with the IMS in separate number registration mode.

Call description:

- When user A calls number 28780020, the AG routes the call to the PBX. Then, the PBX uses the interactive voice response (IVR) function to automatically transfer the call to the desired extension phone, or uses the switchboard to manually transfer the call to the desired extension phone.
- When an extension phone calls user A, the number displayed for user A is 28780020.
- Phone 1 can call phone 30 only using short number 0030 of phone 30.

Figure 1-42 GDN calling



Scenario 3: line hunting calling

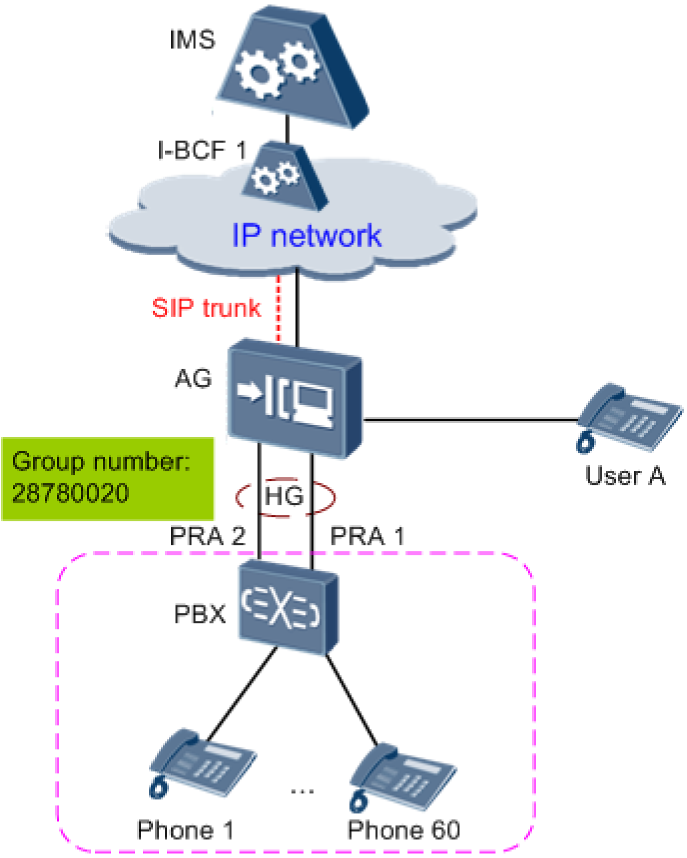
Usage scenario: As shown in Figure 1-43, the enterprise user uses one PBX to connect to 60 extension phones and the PBX connects to the AG using two PRA ports: PRA 1 and PRA 2. The external number of the 60 extension phones is 28780020.

Configuration: PRA 1 and PRA 2 are added to hunting group **HG**, number 28780020 is configured for this hunting group, and the PBX uses this number to register with the IMS.

Call description:

- When user A calls 28780020, the AG routes the call to hunting group **HG** and this group selects an idle PRA port based on hunting policies for call incoming. Then, the PBX uses the IVR function to automatically transfer the call to the desired extension phone, or uses the switchboard to manually transfer the call to the desired extension phone.
- When phone 1 calls user A, the PBX selects an idle PRA port for call outgoing. The number displayed for user A is 28780020.
- Phones 1 through 60 can call each other using only short numbers.

Figure 1-43 Line hunting calling

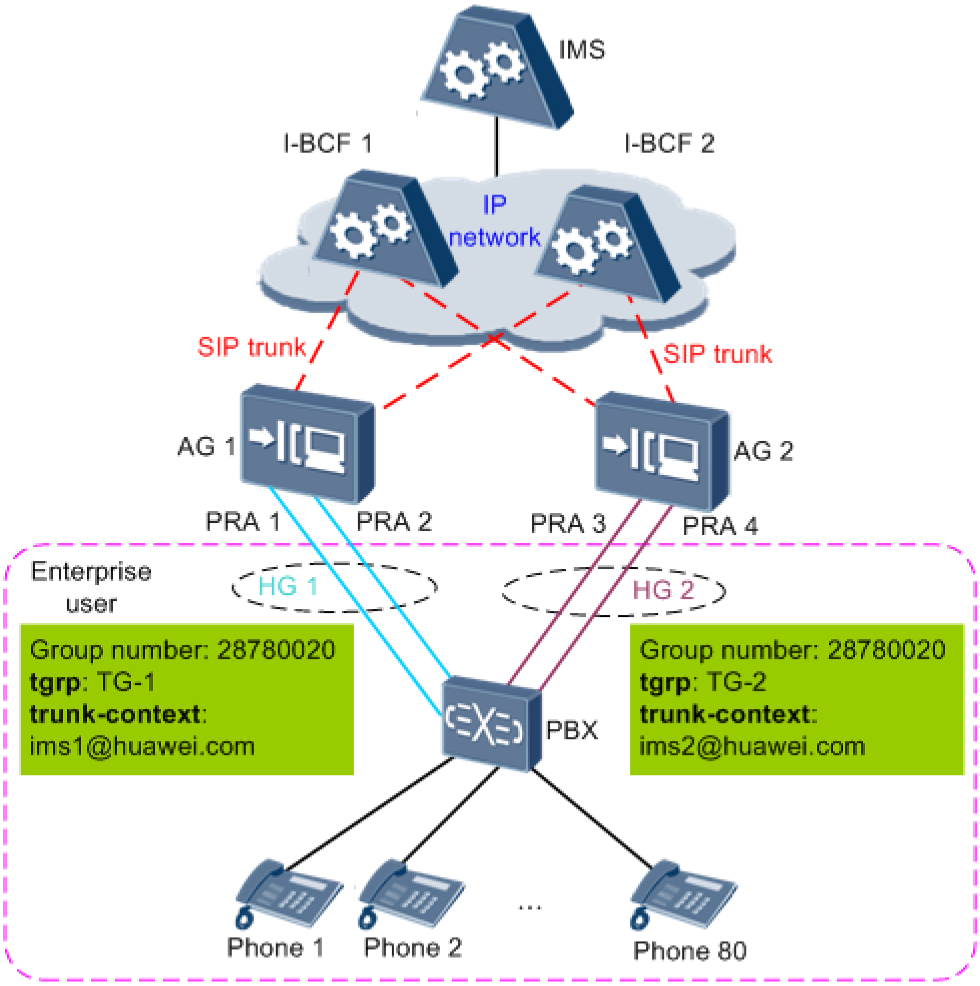


Scenario 4: PBX dual-homing calling (call routing identified by tgrp and trunk-context)

- Usage scenario: As shown in Figure 1-44, the enterprise user uses one PBX to connect to 80 extension phones. The external number of the 80 extension phones is 28780020. Because of high reliability requirements, the PBX uses two PRA ports to connect to two AGs, respectively, for redundancy backup on the access side; both AGs connect to I-BCF 1 and I-BCF 2, respectively, for dual homing on the core network side. This networking improves call reliability. However, the two AGs use the same number. Therefore, calls cannot be routed based on only the phone number. To resolve this issue, **tgrp** and **trunk-context** are added for differentiating between hunting groups for different AGs.
- Configuration:
  - For AG 1:
    - PRA 1 and PRA 2 on AG 1 are added to hunting group **HG1**.
    - Number 28780020 is configured for this hunting group.
    - The values of **tgrp** and **trunk-context** are **TG-1** and **ims1@huawei.com**, respectively.
  - For AG 2:
    - PRA 3 and PRA 4 on AG 2 are added to hunting group **HG2**.
    - Number 28780020 is configured for this hunting group.
    - The values of **tgrp** and **trunk-context** are **TG-2** and **ims2@huawei.com**, respectively.
- Call description:

- When an external user calls number 28780020, the IMS routes the call to the desired AG based on the phone number, **tgrp**, and **trunk-context**. The AG selects an idle PRA port based on hunting policies for call incoming. Then, the PBX uses the IVR function to automatically transfer the call to the desired extension phone, or uses the switchboard to manually transfer the call to the desired extension phone.
- When an extension phone calls an external user, the PBX selects an idle PRA port for call outgoing. The phone number of the calling party is mapped to contact header field URI and carries the **tgrp** and **trunk-context** parameters for the hunting group. Then, the IMS charges the call based on the **tgrp** and **trunk-context** parameters.

Figure 1-44 PBX dual-homing calling



Standards and Protocols Compliance

- RFC 4904: representing trunk groups in tel/sip uniform resource identifiers (URIs)
- RFC 3261: Session Initiation Protocol
- RFC 3966: tel URI for phone numbers
- ETSI TS 182 025: Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); business trunking; architecture and functional description

1.5.3.7 Line Hunting

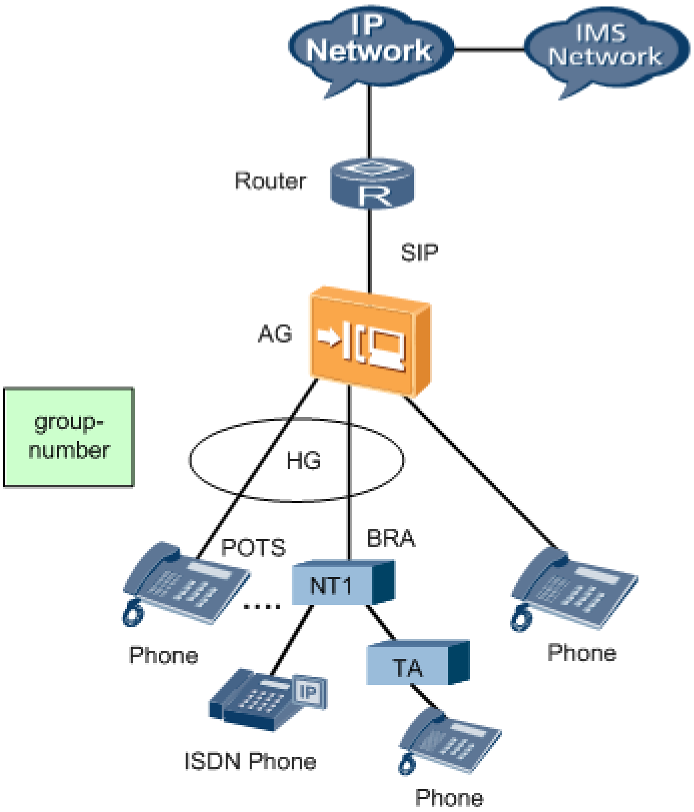
Line hunting is a feature that allows a group of ports to share a group of called party numbers by specifying a hunting group and hunting policy. Only the Session Initiation Protocol (SIP) supports this feature.

Basic Concepts

Figure 1-45 shows a hunting group. The following provides the basic concepts about line hunting.

- Hunting group: a group composed of multiple members that share a group of called party numbers, for example, the HG shown in Figure 1-45
- Group member: a port in a hunting group. A group member can be a POTS port, a BRA port, a PRA port or a child hunting group. A child hunting group is a sub group of a parent hunting group. A child hunting group has the same structure as its parent hunting group. After a child hunting group is added into a parent hunting group, all the members of the child hunting group become the members of the parent hunting group.
- Group number: a called party number shared by members of a hunting group. A group number can be a group numbering reduced (GNR) or a direct dial number (DDN). When the access gateway receives an incoming call and if the incoming call number is a group number of a hunting group, the AG hunts for a group member according to the hunting policy. A hunting group has one or more group numbers, but a group number belongs to only one hunting group.
  - GNR: a prefix of a wildcard number. For example, if the wildcard number is 024545*, the GNR is 024545. For simplicity, "024545*" is called a GNR. If the group number of a hunting group is a GNR, the AG determines that an incoming call number is the group number of the hunting group as long as the first several digits of the incoming call number are identical to the prefix of the GNR.
  - DDN: a specific number instead of a wildcard number
- Hunting policy: a policy used to select a member of a hunting group as the called party for an incoming call. Hunting policies include sequential hunting, circular hunting, and circular hunting by weight. For details, see Hunting Policies.
- Alternative line hunting: defines the call release mode for an ISDN port in a hunting group. Only ISDN ports support alternative line hunting. For details, see Alternative Line Hunting.

Figure 1-45 Hunting group



Hunting Policies

Available hunting policies include sequential hunting, circular hunting, and circular hunting by weight.

Item	Sequential Hunting	Circular Hunting	Circular Hunting by Weight
order	The **order** value determines the priority of a member port. When the AG receives an incoming call, the member port with the **order** value 1 takes precedence. If the member port with the **order** value 1 is busy or faulty, the member port with the **order** value 2 is selected. If the member port with the order value 1 is busy or faulty, the member port with the order value 2 is	The **order** value determines the neighbor relationship between member ports. For example, the ports with the order values 1 and 2 are considered as neighbor ports. The neighbor port next to the previously selected port is the first choice when the AG is hunting for a group member. Specifically, when the AG receives the first incoming call, the member port with the	The meaning is the same as the **order** in circular hunting.

Item	Sequential Hunting	Circular Hunting	Circular Hunting by Weight
	selected, and so on.	**order** value 1 takes precedence. If the port is busy or faulty, the member port with the **order** value 2 is selected. If the member port with the **order** value 2 is successfully selected as the called party, the member port with the **order** value 3 is selected when the AG receives the second incoming call.	
weight	–	–	The **weight** value determines the number of times that a member port is selected as the called party. When the **weight** value of each member port in a hunting group is the same, the circular hunting by weight functions the same as the circular hunting. In circular hunting by weight, the **weight** value decreases by 1 each time after a member port is successfully selected as the called party. If there are still member ports whose **weight** values is not 0 after all member ports in a hunting group are selected once as the called parties, the AG preferentially selects the member ports with non-0 **weight** values as the called parties according to the circular hunting policy.

NOTE

Members of a hunting group can have the same **order** value.

- When a child group and a port have the same **order** value, the port takes precedence.
- When two or more ports have the same **order** value, the ports are selected according their subrack IDs/slot IDs/ port IDs. Specifically, the port in the subrack with the smallest ID takes precedence. If these ports are in the same subrack, the port on the board with the smallest slot ID takes precedence. If these ports are on the same board, the port with the smallest ID takes precedence.
- When two or more sub groups have the same **order** value, they are selected according to the numerical and alphabetical sequences of their names.

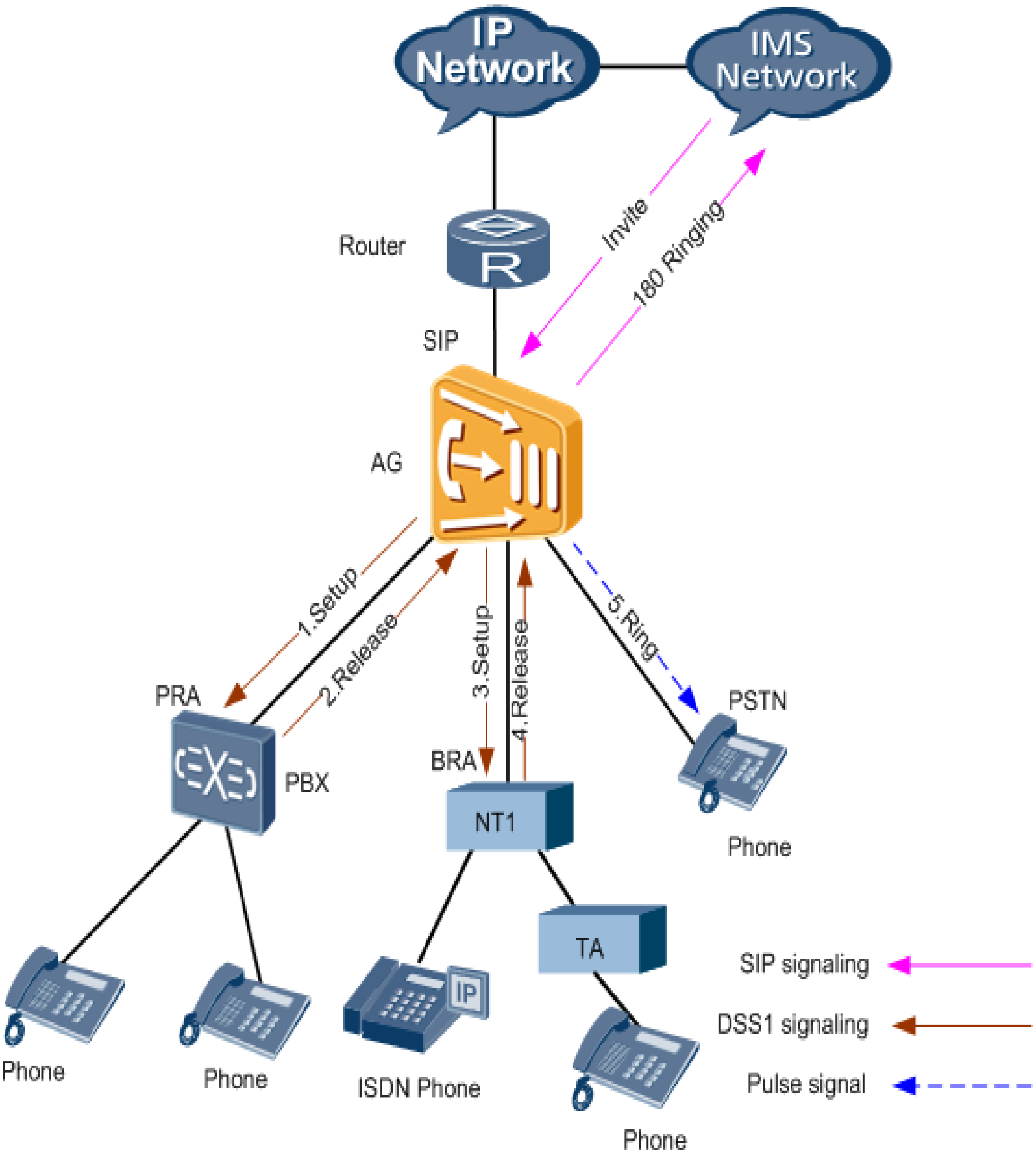
Alternative Line Hunting

Assume that the AG selects an ISDN port as the called party for an incoming call and issues a call request. If the AG receives a call failure message before receiving any response and the Q.850 cause code carried in the call failure message (specifically, Release message) cannot be found in the Q.850 cause code table configured for the hunting group, the AG does not release the call but continues to hunt for another available group member and issues the call request. On the contrary, if the Q.850 cause code is found in the Q.850 cause code table, or if the AG receives a response before receiving any call failure message, the AG releases the call. This procedure is called alternative line hunting. For the definition of the Q.850 cause code, see *ITU-T Q.850*.

Assume that:

- The POTS port, BRA port, and PRA port (as shown in Figure 1-46) are members of the same hunting group.
- The hunting policy of the hunting group is sequential hunting. The **order** value of the PRA port is set to 1. The **order** value of the BRA port is set to 2. The **order** value of the POTS port is set to 3.
- The POTS port, BRA port, and PRA port are in the idle state.
- The Q.850 cause code carried in the Release message cannot be found in the Q.850 cause code table configured for the hunting group.

Figure 1-46 Alternative line hunting



The called party number of an incoming call is a group number. When receiving an Invite message sent by the IP multimedia subsystem (IMS), the AG hunts for a member among multiple group members. The alternative line hunting procedure is as follows:

1. The AG issues a Setup message to the PRA port.
2. The PBX directly replies to the AG with a Release message. The AG cannot find the Q.850 cause code (carried in the Release message) in the Q.850 cause code table configured for the hunting group, and then continues to hunt for a member.
3. The AG issues a Setup message to the BRA port.
4. The BRA replies to the AG with a Release message. The AG cannot find the Q.850 cause code (carried in the Release message) in the Q.850 cause code table configured for the hunting group, and then continues to hunt for a member.

5. The AG issues ringing signals to the POTS port, and replies to the IMS with a 180 Ringing signaling message. Then the POTS port is selected as the called party.

1.5.4 SIP Reference Standards and Protocols

- RFC 3262: Reliability Of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263: SIP Locating SIP Servers

1.6 SIP Value-added Services

SIP value-added services provide more services with easier operations for users and help carriers provide various and flexible services for users. These services improve carriers' competitiveness and user satisfaction.

1.6.1 List of Value-added SIP Services

Table 1-13 describes SIP value-added services.

Table 1-13 SIP value-added services

Service	Definition	POTS User	ISDN User
Call waiting (CW)	If this callee-side service is provisioned and activated and the user who is engaged in an ongoing call is notified of a new incoming call, this new incoming call will be in waiting state and the waiting party hears the CW tone. The user can either accept, reject, or ignore this new incoming call.	Y	Y
Call hold (CH)	The CH service enables a user to temporarily disconnect an established call, that is, to stop sending media streams between the calling party and the called party, but not to release the session resources. When necessary, the call can be resumed. When a user needs to initiate or accept a new call but does not want to release the current call, the user can hold the current call and resume the call when necessary.	Y	Y
Three party conference call (3PTY)	This service provides one call connection for multiple users, that is, it allows 3 users to communicate with each other in the same call.	Y	Y
Conferencing	The conferencing service is a service on the originating side. It provides the multi-connection call capability for a user. That is, multiple users (participants) can participate in the same session at the same time.	Y	Y
Explicit	This service allows user A who is communicating	Y	Y

Service	Definition	POTS User	ISDN User
communication transfer (ECT)	with user B to transfer the call to user C so that a call is set up between user B and user C.		
Call forwarding (CF)	The CF Communication Diversion (CDIV) service is a service on the terminating side. If a user has subscribed to the CF service and the call flow meets the call forwarding conditions, the call is forwarded to the preset forwarded-to party. The system supports the following 3 call forwarding services: call forwarding unconditional (CFU), call forwarding busy (CFB), and call forwarding no reply (CFNR).	Y	Y
Emergency call	A user can make an emergency call if the connected port is in remote block state and the SIP proxy server is normal and the dialed number matches the emergency digitmap.	Y	Y
Hotline	This service functions in this way: If the user who registers this service does not dial a number in a specified period (such as 5s) after picking up the phone, the system automatically connects this user to a fixed number (hotline number).	Y	Y
Message waiting indication (MWI)	This is a message prompt service, with which, the voice mail system (VMS) or unified message system (UMS) notifies users of changed number or status of messages, including emails, short messages, faxes, and leaving messages.	Y	Y
Calling line identification presentation (CLIP)	The CLIP service allows the calling number, name, and other information to be presented to the called party when the called party is alerted.	Y	Y
Calling line identification restriction (CLIR)	The CLIR service enables the calling party to restrict the presentation of the number, name, and multimedia information to the called party during call setup if the calling party has subscribed to the CLIR service and the called party has not subscribed to the RIO service.	Y	Y
Calling line identification restriction override (RIO)	The RIO service is a callee-side service that allows the number of the calling party to be presented to the called party even if the calling party registers CLIR.	Y	Y

Service	Definition	POTS User	ISDN User
Connected line identification presentation (COLP)	The COLP service determines whether to display the called party's identifier to the calling party based on the subscription data of the user and performs corresponding operations.	Y	Y
Callback	When user A calls user B and user B is busy or does not reply, if user A has registered the Completion of Calls to Busy Subscriber (CCBS) or Completion of Communication on No Reply (CCNR) service and the conditions are met, the AG automatically sets up a call when user B is available. The callback service includes the CCBS and CCNR services.	Y	Y
Malicious call identification (MCID)	A user who registers this callee-side service can identify the calling number if the user receives a malicious call or unsolicited call, such as a prank call or a phishing call.	Y	Y
Call hold with three parties	With this service, user A places user B on hold and initiates a new call to user C. Then, user A can switch between the call with user B and the call with user C, or release either of the calls.	Y	Y
Anonymous call service	This service does not allow the number of the calling party who registers this service to be presented to the called party.	Y	Y
Distinctive ringing	A callee-side service, with which, a user can set different ringing tones for different calling parties.	Y	N
Call release control	Call release control is classified into calling party release, called party release, and first party release. They are the 3 modes for call release. • Calling party release: A call is not released if the called party hangs up the phone but the calling party does not. In this case, if the called party picks up the phone again before the timer for calling party release times out, the call is connected and the two parities communicate with each other further. A call is released if the calling party hangs up the phone. • Called party release: A call is not released if the calling party hangs up the phone but the called party does not. In this case, if the calling party picks up the phone again before the timer for called party release times out, the call is connected and the two parities communicate with each other further. A call is released if the called party hangs up the phone.	Y	N

Service	Definition	POTS User	ISDN User
	• First party release: A call is released if either of the calling party and called party hands up the phone.		
Multi-number service	Extended phone numbers are supported for SIP user configurations. The basic number is configured when a user is added or modified. A SIP user supports one basic phone number and multiple extended phone numbers. A SIP user can configure only extended phone numbers (does not configure the basic number). An extended phone number can be a local phone number or global phone number. User registration and subscription can be initiated and managed by phone number. When a SIP user makes a call as caller, the first number functions as the active number. If registration fails, the first number in the following numbers is used for registration again. The rest may be deduced by analogy and the phone number for the first successful registration is bound to the call. When a SIP user makes a call as callee, the SIP access gateway (AG) selects a phone number for the call according to the URI carried in the Invite message.	Y	Y
Advice of charge services	A service that provides the call fees or rate for users. This service can be classified into the following 3 types according to different charging periods. • oAdvice of charge at communication set-up time (AOC-S): provides the rate for users when the rate changes upon the call setup and changes during the communication. **NOTE** For the POTS service, the access node issues the converted charging pulse signal to a user terminal according to the rate. • oAdvice of charge during the communication (AOC-D): provides the call fees for users periodically during the communication. **NOTE** According to ETSI TS 183043, the IMS AS identifies the line type and performs the corresponding operation. For the POTS service, the IMS AS issues the charging information "additional" to the access node. For the ISDN service, the IMS AS issues the charging information "accumulative" to the access node. • oAdvice of charge at the end of the communication (AOC-E): provides the call fees for users of the call just ended after the communication is ended.	Y	Y

1.6.2 Call Waiting

The call waiting (CW) service is callee-side service. If it is provisioned and activated and the user who is engaged in an ongoing call is notified of a new incoming call, this new incoming call will be in waiting state and the waiting party hears the CW tone. The user can either accept, reject, or ignore this new incoming call.

1.6.2.1 Call Waiting

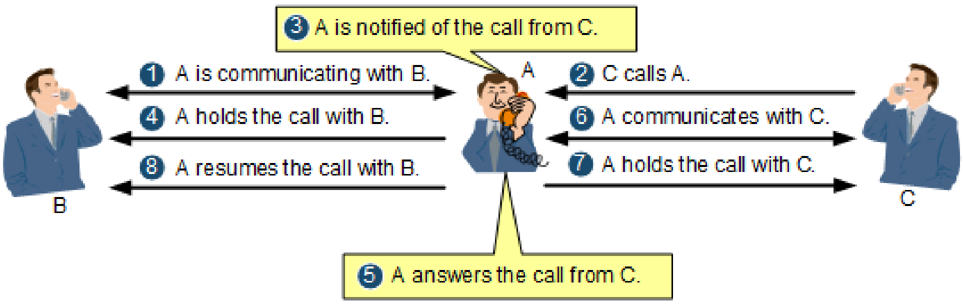
This topic describes the definition and principle of the call waiting (CW) service.

Definition

If this callee-side service is provisioned and activated and the user who is engaged in an ongoing call is notified of a new incoming call, this new incoming call will be in waiting state and the waiting party hears the CW tone. The user can either accept, reject, or ignore this new incoming call.

Assuming that user A registers the CW service, user B is placed on hold, and user C is the third party (CW party), the principle of the CW service is shown in Figure 1-47.

Figure 1-47 Principle of the CW service


3 A is notified of the call from C.
1 A is communicating with B.
A
2 C calls A.
4 A holds the call with B.
6 A communicates with C.
8 A resumes the call with B.
7 A holds the call with C.
B
C
5 A answers the call from C.

1. User A is communicating with user B.
2. User C calls user A.
3. User A receives the incoming call prompt from user C.
4. User A wants to communicate with user C and places user B on hold.
5. User A answers the call from user C.
6. User A communicates with user C.
7. User A presses hookflash to hold the call with user C.
8. User A restores the call with user B.

Benefit

Beneficiary	Benefits
Carrier	The CW service helps improve the ratio of successful call connections, enhances carriers' competitiveness and provides carriers with new revenue

Beneficiary	Benefits
	growth potential.
Users	- The CW service prevents a caller from dialing a number for multiple times and improves the ratio of successful call connections. - The CW service reduces the number of missed calls and the possibility of missing expected calls. The CW service can be canceled temporarily so that the ongoing call is not affected.

Standards Compliance

- ETSI TS 03027
- 3GPP TS 24615-820
- ETSI EN 300 058-1
- ETSI 300 102-1
- ETSI TS 183 015
- ETSI TS 186 022
- ETSI TS 183 036
- ETSI TS 183 043
- ITU-T Q.931

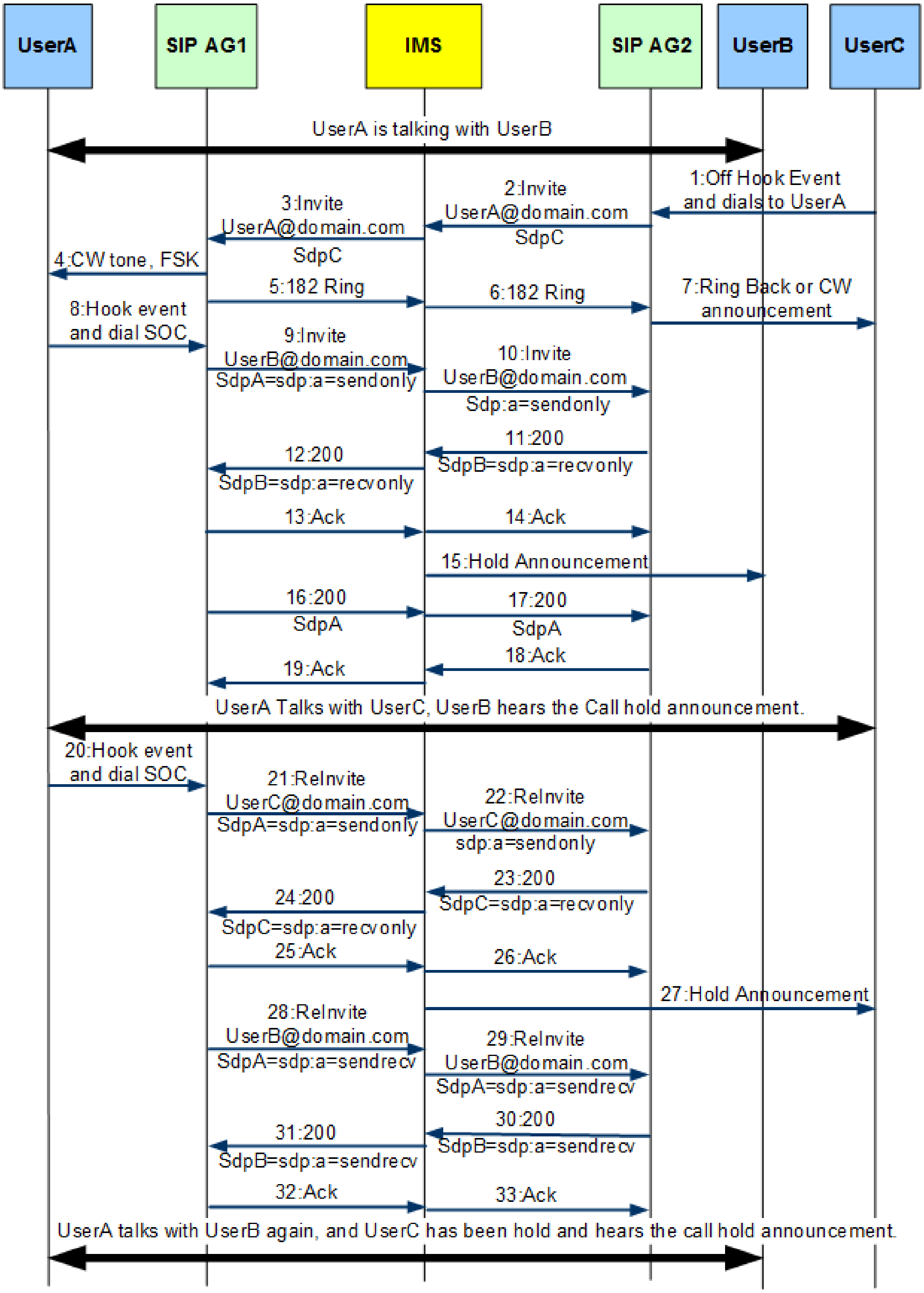
1.6.2.2 Call Waiting Service Flow

This topic takes users A, B, and C as an example to describe the service flow of call waiting for a POTS and ISDN user.

POTS Call Waiting Service Flow

Figure 1-48 shows the service flow of the POTS call waiting service. User A is communicating with user B, user A places user B on hold and communicates with user C (from waiting to connected), and then user A puts user C on hold and communicates with user B (from on hold to connected).

Figure 1-48 POTS call waiting service flow


UserA
SIP AG1
IMS
SIP AG2
UserB
UserC
UserA is talking with UserB
1:Off Hook Event and dials to UserA
2:Invite UserA@domain.com SdpC
3:Invite UserA@domain.com SdpC
4:CW tone, FSK
5:182 Ring
6:182 Ring
7:Ring Back or CW announcement
8:Hook event and dial SOC
9:Invite UserB@domain.com SdpA=sdp:a=sendonly
10:Invite UserB@domain.com Sdp:a=sendonly
11:200 SdpB=sdp:a=recvonly
12:200 SdpB=sdp:a=recvonly
13:Ack
14:Ack
15:Hold Announcement
16:200 SdpA
17:200 SdpA
18:Ack
19:Ack
UserA Talks with UserC, UserB hears the Call hold announcement.
20:Hook event and dial SOC
21:ReInvite UserC@domain.com SdpA=sdp:a=sendonly
22:ReInvite UserC@domain.com sdp:a=sendonly
23:200 SdpC=sdp:a=recvonly
24:200 SdpC=sdp:a=recvonly
25:Ack
26:Ack
27:Hold Announcement
28:ReInvite UserB@domain.com SdpA=sdp:a=sendrecv
29:ReInvite UserB@domain.com SdpA=sdp:a=sendrecv
30:200 SdpB=sdp:a=sendrecv
31:200 SdpB=sdp:a=sendrecv
32:Ack
33:Ack
UserA talks with UserB again, and UserC has been hold and hears the call hold announcement.

The call waiting service flow for a POTS user is as follows:

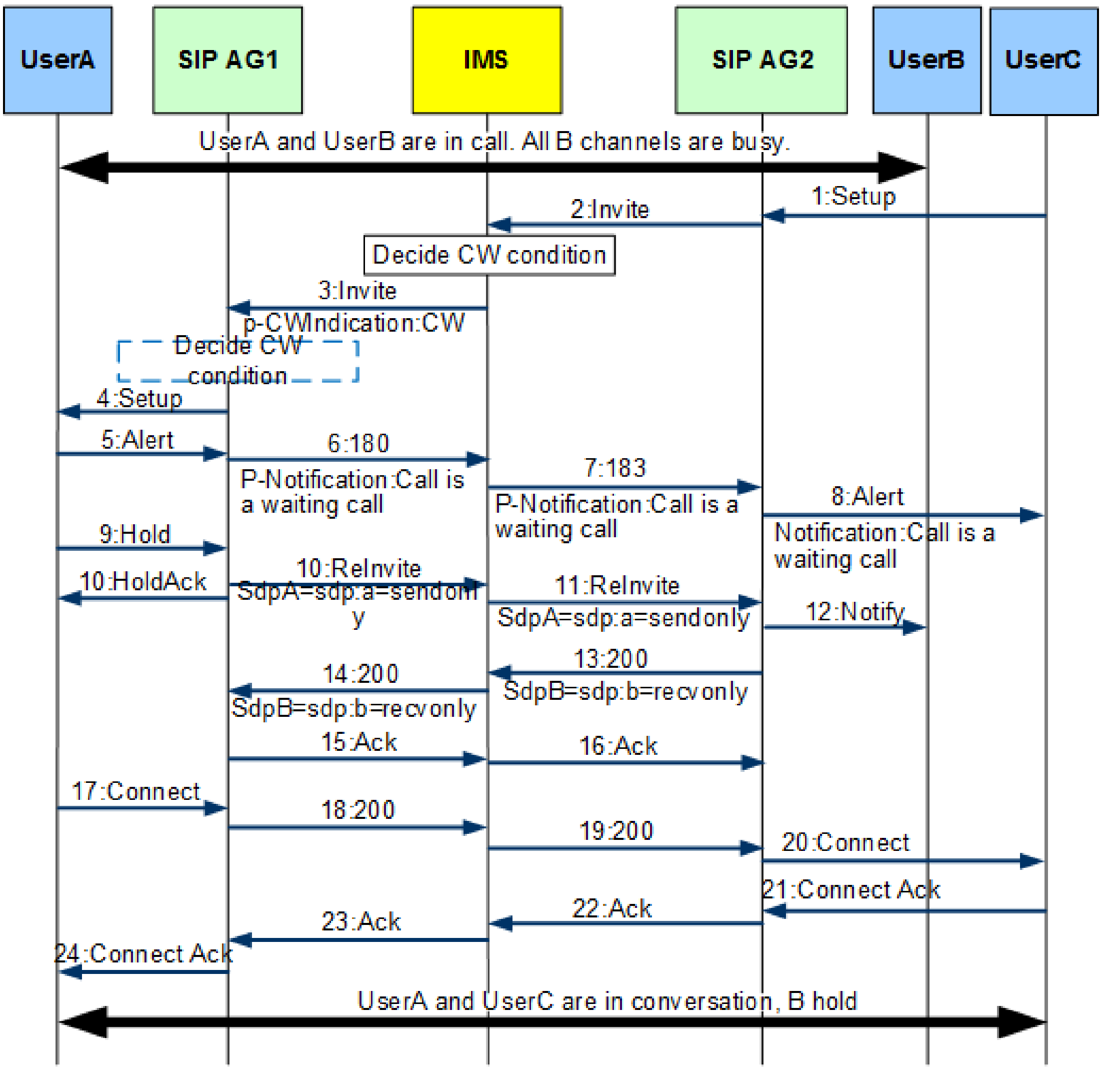
1. User A is communicating with user B. User C picks up the phone and calls user A.
2. AG2 sends an Invite message to user A.
3. The IMS sends an Invite message to AG1.
4. After receiving the call message from user C, AG1 determines that A has the call waiting service permission and plays the CW tone to A.
5. AG1 sends a 182 response to the IMS.
6. The IMS sends a 182 response to AG2.
7. After receiving the 182 (or 180) response from the IMS, AG2 plays the ringback tone or call waiting tone to C.
8. User A presses hookflash to answer the call from user C, hears the special dial tone, and dials 2 (SOC).
9. After detecting the hookflash event, AG1 sends an Invite message to user B, setting the media stream direction to sendonly.
10. After receiving the Invite message, the IMS sends the Invite message to AG2.
11. After receiving the message, AG2 modifies the media attribute and direction of user B and returns a 200 OK response to the IMS. The 200 OK response carries the SDP, indicating that user B can only receive media.
12. The IMS sends the 200 OK response to AG1.
13. After receiving the 200 OK message, AG1 sends an Ack message to the IMS.
14. The IMS sends an Ack message to AG2.
15. User B is placed on hold and hears a call hold tone.
16. AG1 returns a 200 OK response to user C. The 200 OK response carries an SDP packet, indicating that user A can receive and send media.
17. The IMS sends a 200 OK response to AG2.
18. After receiving the 200 OK message, AG2 sends an Ack message to the IMS.
19. The IMS sends an Ack message to AG1. The call with user B is held successfully, and user B hears a call hold tone. User A starts communicating with user C.
20. User A presses hookflash to communicate with user B, hears the special dial tone, and dials 2 (SOC).
21. AG1 sends a ReInvite message to user C, with an SDP setting the media direction to sendonly.
22. The IMS sends the ReInvite message to AG2.
23. After receiving the ReInvite message, AG2 changes the media direction to recvonly and returns a 200 response to the IMS.
24. The IMS sends a 200 message to AG1.
25. AG1 sends an Ack message to the IMS.
26. The IMS sends an Ack message to AG2.
27. After receiving the Ack message, AG2 plays the call hold tone to user C and the call with user C is held successfully.
28. After user C is held, AG1 sends a ReInvite message (in which the SDP direction is sendrecv) to resume the call with user B.
29. The IMS receives the ReInvite message and sends the ReInvite message to AG2.
30. After receiving the ReInvite message, AG2 changes the media direction to sendrecv and returns a 200 message to the IMS.

31. The IMS sends the 200 message to user A.
32. After receiving the 200 message, AG1 sends an Ack message to the IMS.
33. The IMS sends an Ack message to AG2. The call between users A and B is resumed.

CW Service Flow for an ISDN User

The CW service flow for ISDN users is implemented in either of the following cases based on whether the service detection is performed on the IMS or ATOM GPS, as shown in Figure 1-49.

Figure 1-49 ISDN user CW service flow-IMS detection



The CW service flow for an ISDN user is as follows:

1. User A is communicating with non-service user B. Non-service user C initiates a call to user A.
2. AG2 initiates a call to user A.
3. When the IMS determines that the number of calls and media of the called party exceeds the maximum number of calls and media, and no other services conflict with this call.

That is, the CW service is triggered if the CW conditions are met. The IMS sends an Invite message to AG1. The Invite message carries the CW-Indication indicator.

4. After receiving the Invite message that carries the CW-Indication indicator, AG1 determines that the CW service should be triggered by the IMS and maps the Setup message to the service party.
5. User A answers the Alert message.
6. AG1 sends a 180 response to the IMS and adds the P-Notification:Call is a waiting call header field to indicate the call waiting status.
7. The IMS sends a 183 response to AG2.
8. After receiving the 183 response, AG2 maps the 183 response to the Alert message of user C. The message carries the notification information element, telling user C: "Call is a waiting call."
9. User A presses hookflash to place user B on hold.
10. After detecting the hookflash event, AG1 converts the event into a ReInvite message and returns a user A Hold Ack response message.
11. After receiving the ReInvite message, the IMS modifies the SDP and sends a ReInvite message to user B. After receiving the ReInvite message, AG2 modifies the media attribute of user B so that user B can only receive but cannot send media.
12. AG2 sends a Notify message to notify user B that user B is held.
13. AG2 sends a 200 OK response to the IMS.
14. The IMS sends a 200 OK response.
15. After receiving the 200 OK message, AG1 sends an Ack message to user B.
16. The IMS sends an Ack message to user B.
17. User A sends a Connect message for communicating with user C.
18. After receiving the Connect message, AG1 maps the Connect message to the 200 message and sends the 200 message to the IMS.
19. The IMS sends a 200 message to AG2.
20. After receiving the 200 message, AG2 sends a Connect message to user C.
21. After receiving the Connect message, user C returns a ConnectAck message.
22. AG1 sends an Ack message to the IMS.
23. The IMS sends an Ack message to AG2.
24. User A receives a ConnectAck message. The call between users A and C is set up.

NOTE

- The preceding flow is the call waiting service detection flow on the IMS side. Another scenario involves detection on the AG side, as shown in the blue dashed line box in Figure 1-49. The difference between detection on the AG side and detection on the IMS side in the call waiting service flow for ISDN users is as follows: The Invite message sent from the IMS to AG1 does not carry the CW-Indication indicator. AG1 determines whether to trigger the CW service based on whether the called party has the CW service permission and whether the B channel is unavailable. The other steps are the same and are not described here.
- When A has a call waiting message on the ISDN terminal, the following options are available: 1. to accept the waiting party and hold the current party; 2. to accept the waiting party and release the current party; 3. to reject the waiting party and continue the current call. The preceding service flow uses the first option as an example. The flow of accepting the waiting party, releasing the current party, and releasing the waiting party is the same as that of a basic call service. For details, see the corresponding service.

1.6.3 Call Hold

The call hold (CH) service enables a user to temporarily disconnect an established call, that is, to stop sending media streams between the calling party and the called party, but not to release the session resources. When necessary, the call can be resumed. When a user needs to initiate or accept a new call but does not want to release the current call, the user can hold the current call and resume the call when necessary.

1.6.3.1 Call Hold

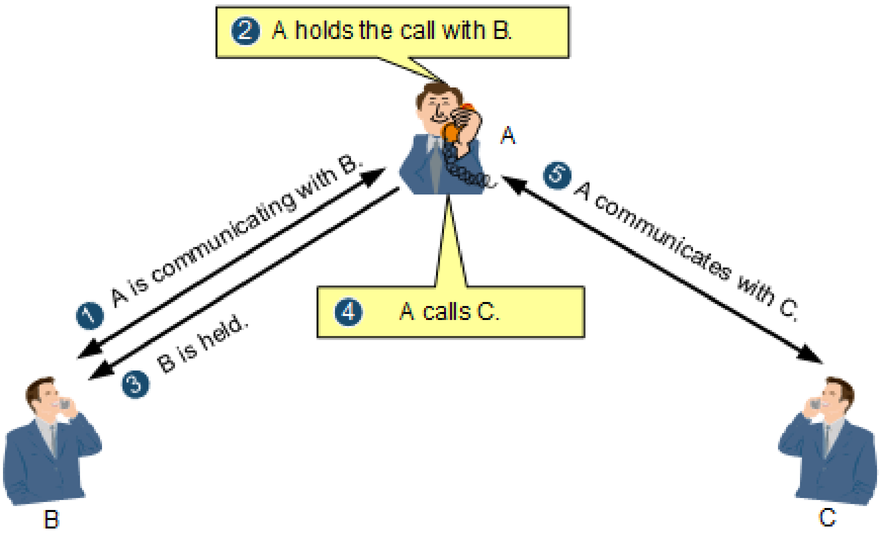
This topic describes the definition and principle of the call hold (CH) service.

Definition

The CH service enables a user to temporarily disconnect an established call, that is, to stop sending media streams between the calling party and the called party, but not to release the session resources. When necessary, the call can be resumed. When a user needs to initiate or accept a new call but does not want to release the current call, the user can hold the current call and resume the call when necessary.

User A has subscribed to the CH service, user B is the held party, and user C is the third party, as shown in Figure 1-50.

Figure 1-50 Principle of the CH service



1. User A is communicating with user B.
2. User A wants to communicate with user C and presses hookflash to hold the call.
3. User B is placed on hold and hears a call hold tone.
4. User A dials user C's phone number.
5. User C answers the call and communicates with user A.

NOTE
The preceding figure shows one of the application scenarios. When C calls A, A can hold B and answer the call from C.

Benefit

Beneficiary	Benefits
Carrier	The CH service supplements the value-added services of carriers and helps improve the ratio of successful call connections.
Users	A user can place the ongoing call on hold and then resumes this call if required, reducing the number of dials and facilitating call making.

Standards Compliance

- EN300 141
- ETSI TS 124 410
- ETSI 300 102-1
- ETSI TS 183 010
- ETSI TS 183 036
- ETSI TS 183 043
- ITU-T Q.931

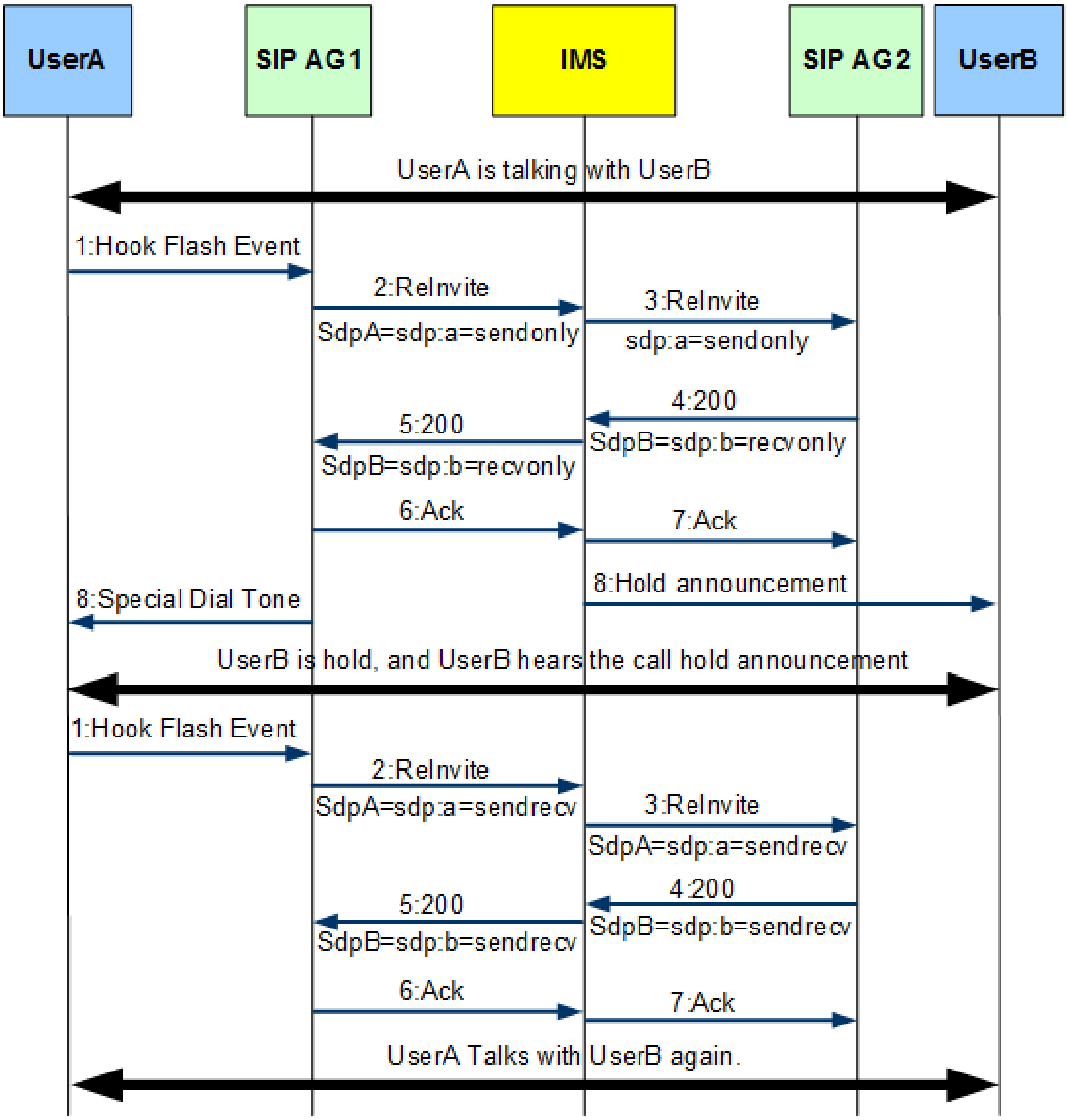
1.6.3.2 Call Hold Service Flow

This topic takes users A and B as an example to describe the service flows of call hold and hold resume.

POTS Call Hold and Hold Resume Service Flow

Figure 1-51 shows the service flow of the POTS call hold and hold resume service.

Figure 1-51 POTS call hold and hold resume service flow



The service flow of call hold for POTS user A is as follows:

1. User A is communicating with user B, and user A presses hookflash.
2. After detecting the hookflash event, AG1 converts the event into a ReInvite message and sends a request to user B.
3. After receiving the ReInvite message, the IMS modifies the SDP and sends a ReInvite message to user B. After receiving the ReInvite message, AG2 modifies the media attribute of user B. User B receives but cannot send media.
4. AG2 returns a 200 OK response carrying an SDP packet, indicating that user B only receives media.
5. The IMS sends a 200 OK response, and AG1 receives the 200 OK message.
6. AG1 sends an Ack message to user B.
7. The IMS sends an Ack message to user B.

8. User A hears a mute or special dialing tone. User B is placed on hold and hears a call hold tone.

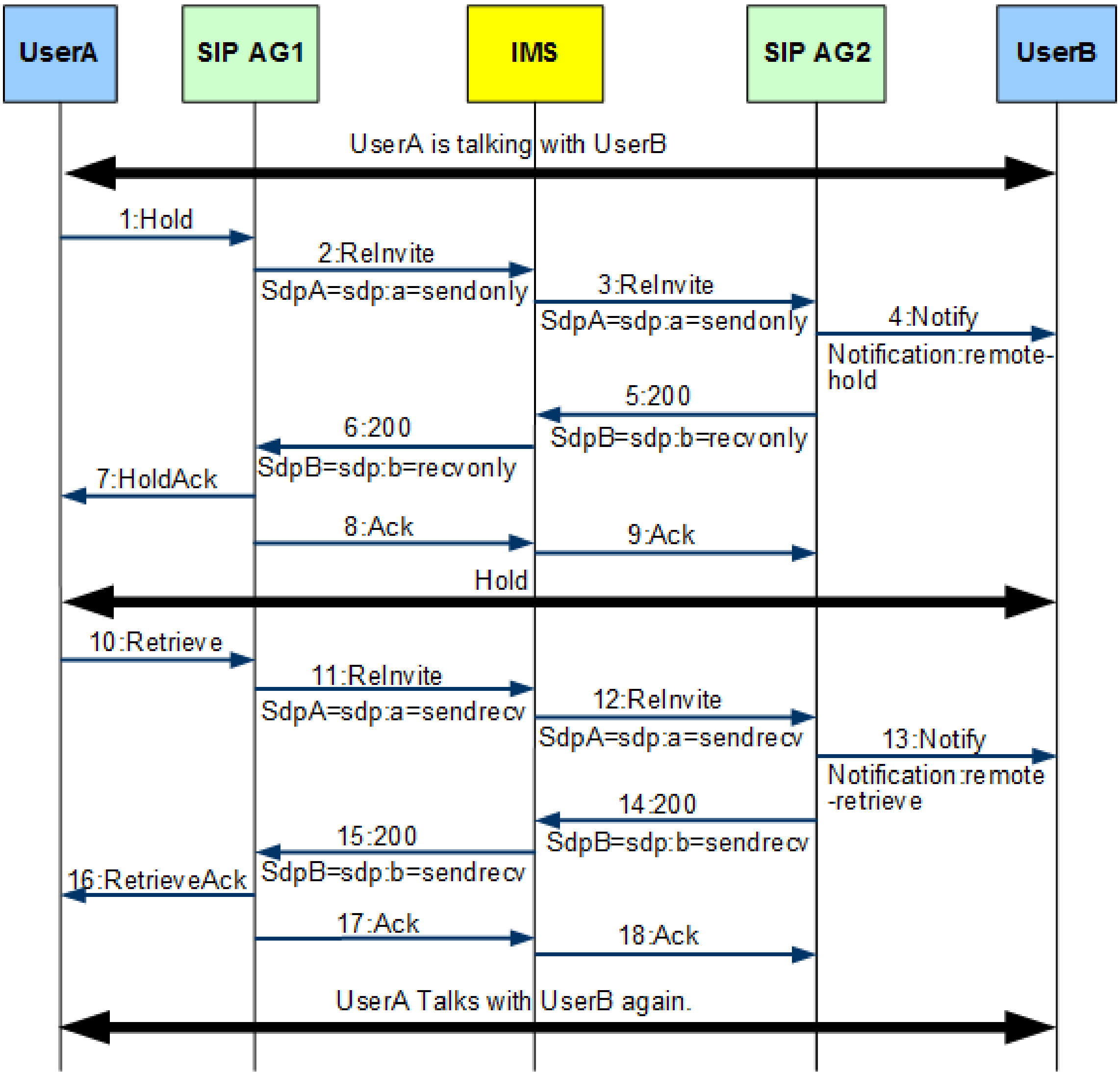
The call release flow for POTS user A is as follows:

1. User B is on hold and A presses hookflash.
2. After detecting the hookflash event, AG1 converts the event into a ReInvite message and sends a request to the IMS.
3. After receiving the ReInvite message, the IMS sends a ReInvite message to user B. After receiving the ReInvite message, AG2 modifies the media attribute of user B. User B can receive and send media.
4. AG2 returns a 200 OK response which carries an SDP packet, indicating that user B can receive and send media.
5. AG1 receives the 200 OK response from the IMS.
6. AG1 sends an Ack message to user B.
7. The IMS sends an Ack message to user B, and AG2 receives the Ack message. The call between users A and B is resumed.

Service Flow of the ISDN Call Hold and Hold Resume Service

Figure 1-52 shows the service flow of the ISDN call hold and hold resume service.

Figure 1-52 Service flow of the ISDN call hold and hold resume service



The service flow is as follows:

1. User A sends a Hold message.
2. AG1 converts the Hold message to a ReInvite message.
3. The IMS applies for multimedia resource function processor (MRFP) resources for announcement playing, modifies SDP information, and sends a ReInvite message to the called party.
4. AG2 converts the ReInvite message into an ISDN Notify message, modifies the media attribute, and uses the Notify message to notify the remote hold.
5. User B is placed on hold, and AG2 returns a 200 message.
6. The IMS sends a 200 response.
7. After receiving the 200 response, AG1 sends a HoldAck message to the service terminal.
8. AG1 sends an Ack message to the IMS.
9. The IMS sends an Ack message to AG2.

NOTE

Steps 10 to 18 are the same as the call hold flow for ISDN users. The difference lies in that the flow mode of user B is **sendrecv**.

1.6.4 Three Party Conference Call

This service provides one call connection for multiple users, that is, it allows 3 users to communicate with each other in the same call.

1.6.4.1 Three Party Conference Call

This topic describes the definition and principle of the three party conference call (3PTY) service.

Definition

This service provides one call connection for multiple users. That is, it allows 3 users to communicate with each other in the same call.

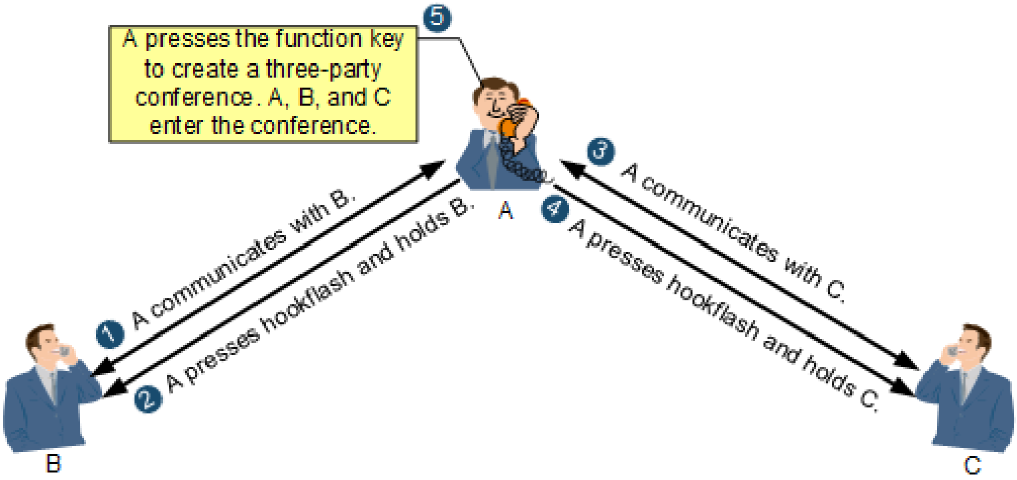
Based on whether 3PTY call resources are allocated locally or allocated by the IMS, the 3PTY service is classified into local 3PTY and remote 3PTY.

The service user can use the 3PTY service in the following ways:

- The service user can create a 3PTY conference and call the other two users at the same time. After the other two users answer the call, the three users communicate with each other.
- The service party can call a user first, and then call the third user after a call is set up. After the third user answers the call, the three users communicate with each other.

User A subscribes to the 3PTY service, users B and C are common users. User A creates a 3PTY call with users B and C in the second manner, as shown in Figure 1-53.

Figure 1-53 Principle of the 3PTY service


5
A presses the function key to create a three-party conference. A, B, and C enter the conference.
1 A communicates with B.
2 A presses hookflash and holds B.
A
3 A communicates with C.
4 A presses hookflash and holds C.
B
C

1. User A calls and communicates with user B.
2. User A presses hookflash to place user B on hold. Then, user B hears a call hold tone.
3. User A calls and communicates with user C.

4. User A presses hookflash to place user C on hold. Then, user C hears a call hold tone.
5. User A presses the function key to set up a 3PTY call. Then, users A, B, and C communicate with each other. Generally, the function key is 3.

NOTE

Generally, the function key is 3.

Benefit

Beneficiary	Benefits
Carrier	The 3PTY service supplements the value-added services of carriers, provides multi-party calls for users, and improves user satisfaction.
User	The 3PTY service enables a user to initiate a 3PTY call for discussion, facilitating user communication and improving communication efficiency.

Standards Compliance

- EN300 188
- ETSI 300 102-1
- ETSI TS 183 005
- 3GPP TS 24147
- 3GPP TS 24605
- 3GPP TS 24610
- 3GPP TS 23218-630
- ETSI TS 183 036
- ETSI TS 183 043
- ITU-T Q.931

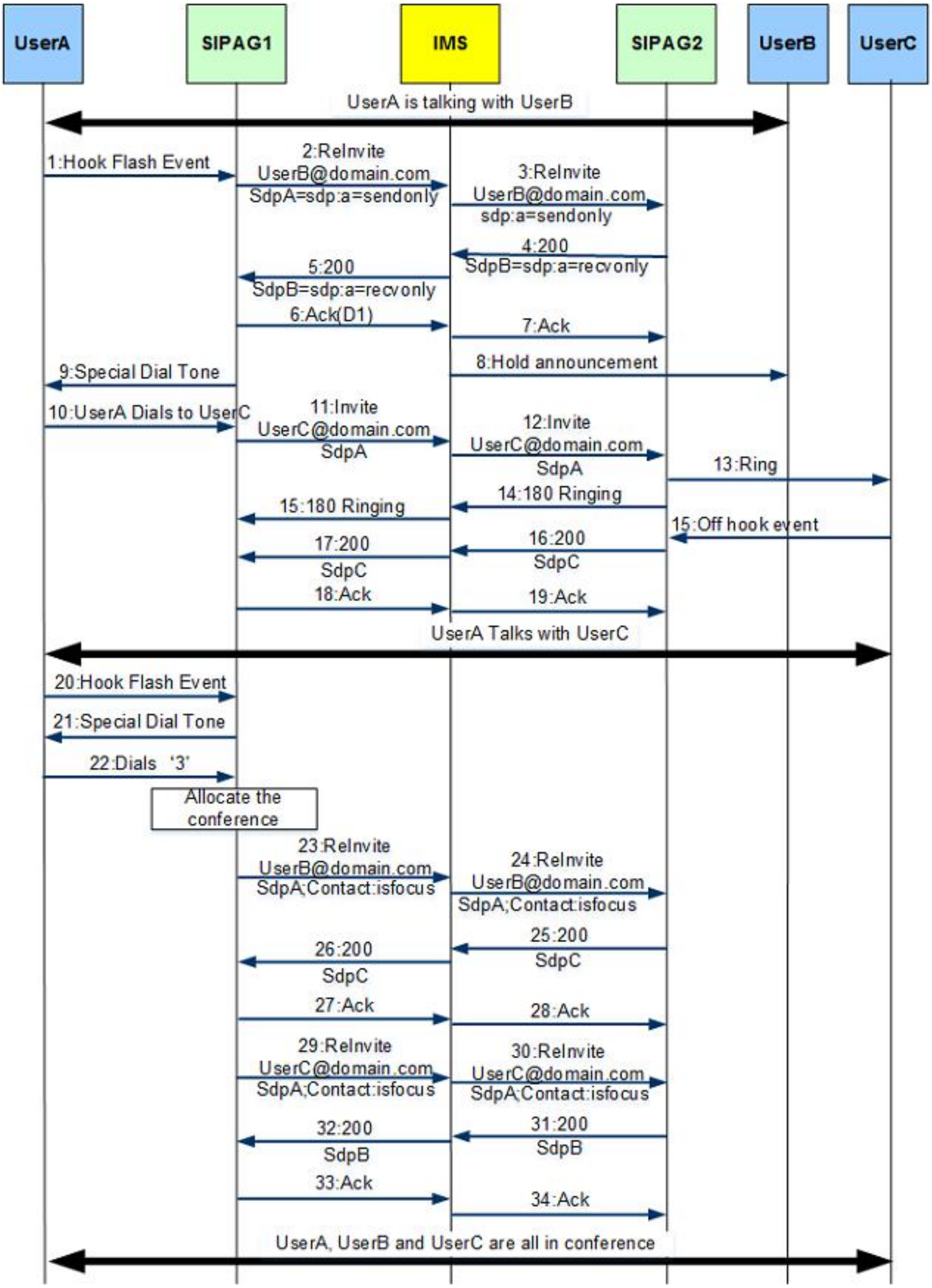
1.6.4.2 Service Flow of Local Three Party Conference Call

In the following service flows of local three party conference call (3PTY) for POTS users or ISDN users, user A is the one who subscribes to the service, and users B and C are common users.

Local 3PTY for POTS Users

Figure 1-54 shows how local 3PTY for POTS users is implemented.

Figure 1-54 Service flow of local 3PTY for POTS users


UserA
SIPAG1
IMS
SIPAG2
UserB
UserC
UserA is talking with UserB
1:Hook Flash Event
2:ReInvite UserB@domain.com SdpA=sdp:a=sendonly
3:ReInvite UserB@domain.com sdp:a=sendonly
4:200 SdpB=sdp:a=recvonly
5:200 SdpB=sdp:a=recvonly
6:Ack(D1)
7:Ack
8:Hold announcement
9:Special Dial Tone
10:UserA Dials to UserC
11:Invite UserC@domain.com SdpA
12:Invite UserC@domain.com SdpA
13:Ring
14:180 Ringing
15:180 Ringing
15:Off hook event
16:200 SdpC
17:200 SdpC
18:Ack
19:Ack
UserA Talks with UserC
20:Hook Flash Event
21:Special Dial Tone
22:Dials '3'
Allocate the conference
23:ReInvite UserB@domain.com SdpA;Contact:isfocus
24:ReInvite UserB@domain.com SdpA;Contact:isfocus
25:200 SdpC
26:200 SdpC
27:Ack
28:Ack
29:ReInvite UserC@domain.com SdpA;Contact:isfocus
30:ReInvite UserC@domain.com SdpA;Contact:isfocus
31:200 SdpB
32:200 SdpB
33:Ack
34:Ack
UserA, UserB and UserC are all in conference

The service flow of local 3PTY for POTS users is as follows:

1. When user A talks with user B, user A presses hookflash.
2. AG1 detects the hookflash pressing event, converts the event into a ReInvite message, and sends the message to user B.
3. After the IMS receives the ReInvite message, it applies for announcement playing resources and sends a ReInvite message to user B. After AG2 receives a ReInvite message, AG2 modifies the media attribute of user B so that user B can receive but cannot send media streams.
4. AG2 replies with a 200 OK response carrying an SDP packet, indicating that user B only receives media streams.
5. The IMS sends a 200 OK response, and AG1 receives the 200 OK message.
6. AG1 sends an Ack message to user B.
7. The IMS sends an Ack message to user B, and AG2 receives the Ack message.
8. User B is successfully placed on hold and hears a call hold tone.
9. User A hears a special dial tone or hears nothing.
10. User A dials user C's number.
11. AG1 sends an Invite message to user C after detecting the dialing event.

NOTE

In steps 11 to 19, AG1 initiate a call to user C. The flow is the same as a basic call flow.

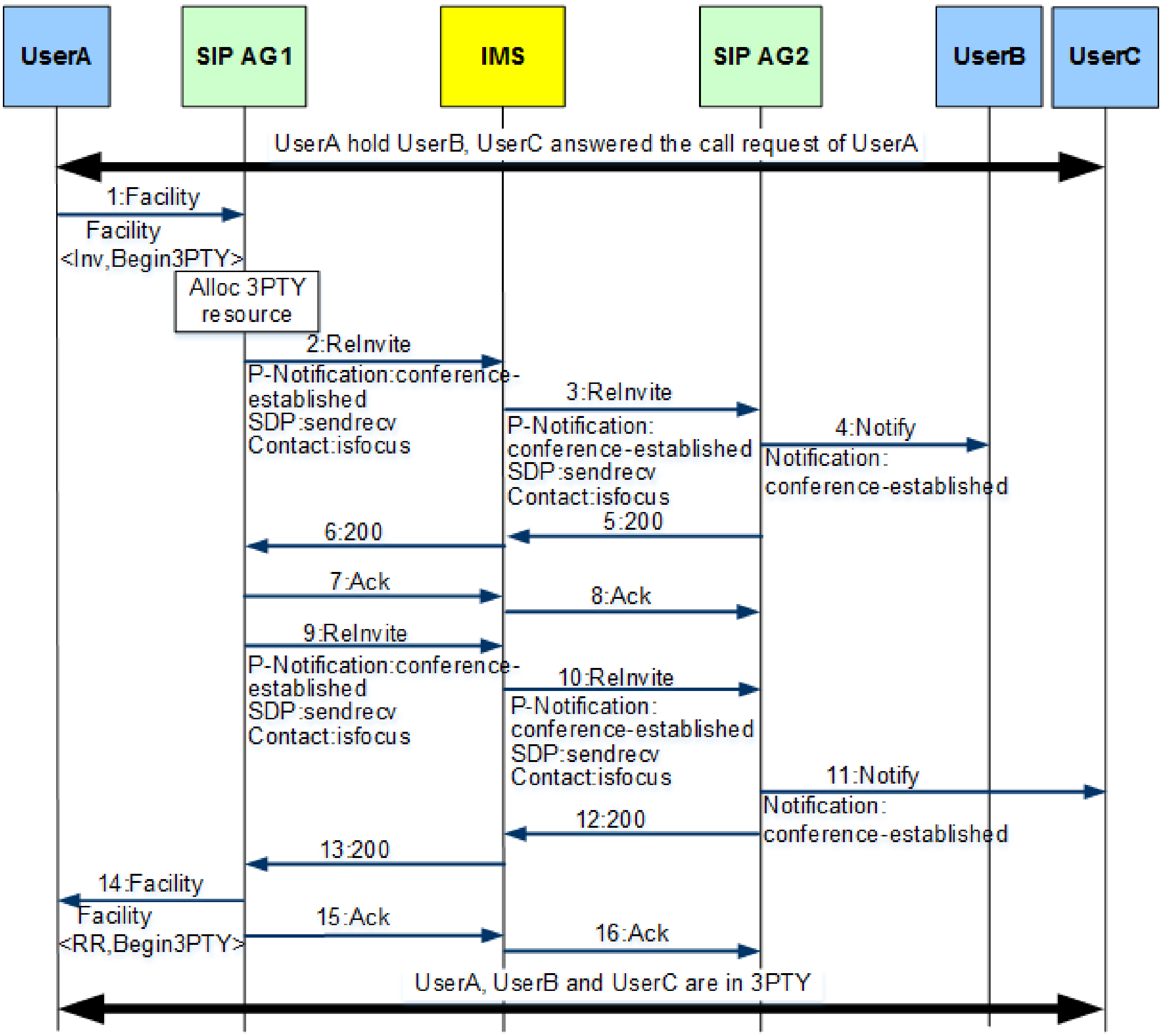
12. After the IMS receives the Invite message, it sends the Invite message to user C. AG2 receives the Invite message.
13. AG2 alerts user C.
14. AG2 sends a 180 response to AG1.
15. The IMS sends the 180 response sent by AG2. User C picks up the phone after hearing a ringing tone.
16. AG2 sends a 200 OK message.
17. The IMS sends the 200 OK message sent by AG2.
18. AG1 sends an Ack response.
19. The IMS sends an Ack message to user B, and AG2 receives the Ack message. Then a call is set up between users A and C.
20. User A presses hookflash again.
21. After pressing hookflash, user A hears a special dial tone.
22. User A dials service code "3" in an attempt to participate in a three-party conference.
23. After AG1 detects the hookflash event and service code "3", AG1 locally applies for conference resources, converts data into a ReInvite message carries **isfocus** parameter data, and sends the message to the IMS, asking the IMS to add B to the conference.
24. After the IMS receives the ReInvite message, it sends the ReInvite message to user B. After AG2 receives the ReInvite message, AG2 modifies the media attribute and direction of user B to **sendrecv**.
25. AG2 sends a 200 OK response to the IMS. The 200 OK response carries an SDP packet, instructing that user B can receive and send media streams.
26. The IMS sends a 200 OK response to user A connected to AG1.
27. After AG1 receives the 200 OK message, it sends an Ack message to user B.
28. The IMS sends an Ack message to user B, and AG2 receives the Ack message.

29. AG1 sends a ReInvite message (the SDP direction is **sendrecv**), and user C is added to the conference.
30. After the IMS receives the ReInvite message, it sends the ReInvite message to user C connected to AG2.
31. After AG2 receives the ReInvite message, AG2 modifies the media direction and sends a 200 response to C.
32. AG1 receives the 200 message that is sent from the IMS to user C.
33. After AG1 receives the 200 message that is sent from the IMS to user C, AG1 sends an Ack message to the IMS.
34. After user C connected to AG2 receives the Ack message, user C is added to the conference. Then all the three parties (A, B, and C) are in the conference.

Local 3PTY for ISDN Users

Figure 1-55 shows how local 3PTY for ISDN users is implemented.

Figure 1-55 Service flow of local 3PTY for ISDN users



The service flow of local 3PTY for ISDN users is as follows:

1. User A places user B on hold and sets up a call with user C. User A initiates a three-party call on an ISDN terminal. That is, user A sends a Facility message to AG1 through CR1.
2. AG1 locally applies for three-party conference room resources. To make user B enter a conference room for normal conversation, AG1 sends a ReInvite message to user B and modifies the stream mode to **sendrecv**.
3. After the IMS receives the ReInvite message, it modifies SDP data and sends the ReInvite message to user B. After AG2 receives the ReInvite message, AG2 modifies the media attribute and direction of user B to **sendrecv**.
4. After AG2 receives the message, it converts the message into a Notify message to notify user B.
5. The AG to which user B is connected sends a 200 OK message.
6. The IMS sends a 200 OK response, and AG1 receives the 200 OK message.
7. AG1 sends an Ack message to user B.
8. The IMS sends an Ack message to user B, and AG2 receives the Ack message.
9. Similarly, AG1 invites user C to the conference room. AG1 sends a ReInvite message to user C. Because a bidirectional call is already set up between users A and C, the flow mode does not need to be modified.
10. After the IMS receives the ReInvite message, it sends the ReInvite message to user C. AG2 receives the ReInvite message.
11. After AG2 receives the message, it converts the message into a Notify message to notify user C.
12. The AG to which user C is connected sends a 200 OK message.
13. The IMS sends a 200 OK response, and AG1 receives the 200 OK message.
14. After AG1 receives the ReInvite response, AG1 sends a Facility message to user A.
15. AG1 sends an Ack message to user C.
16. The IMS sends an Ack message to user C. The AG to which user C is connected receives the Ack message. Then users A, B, and C successfully join the three-party conference.

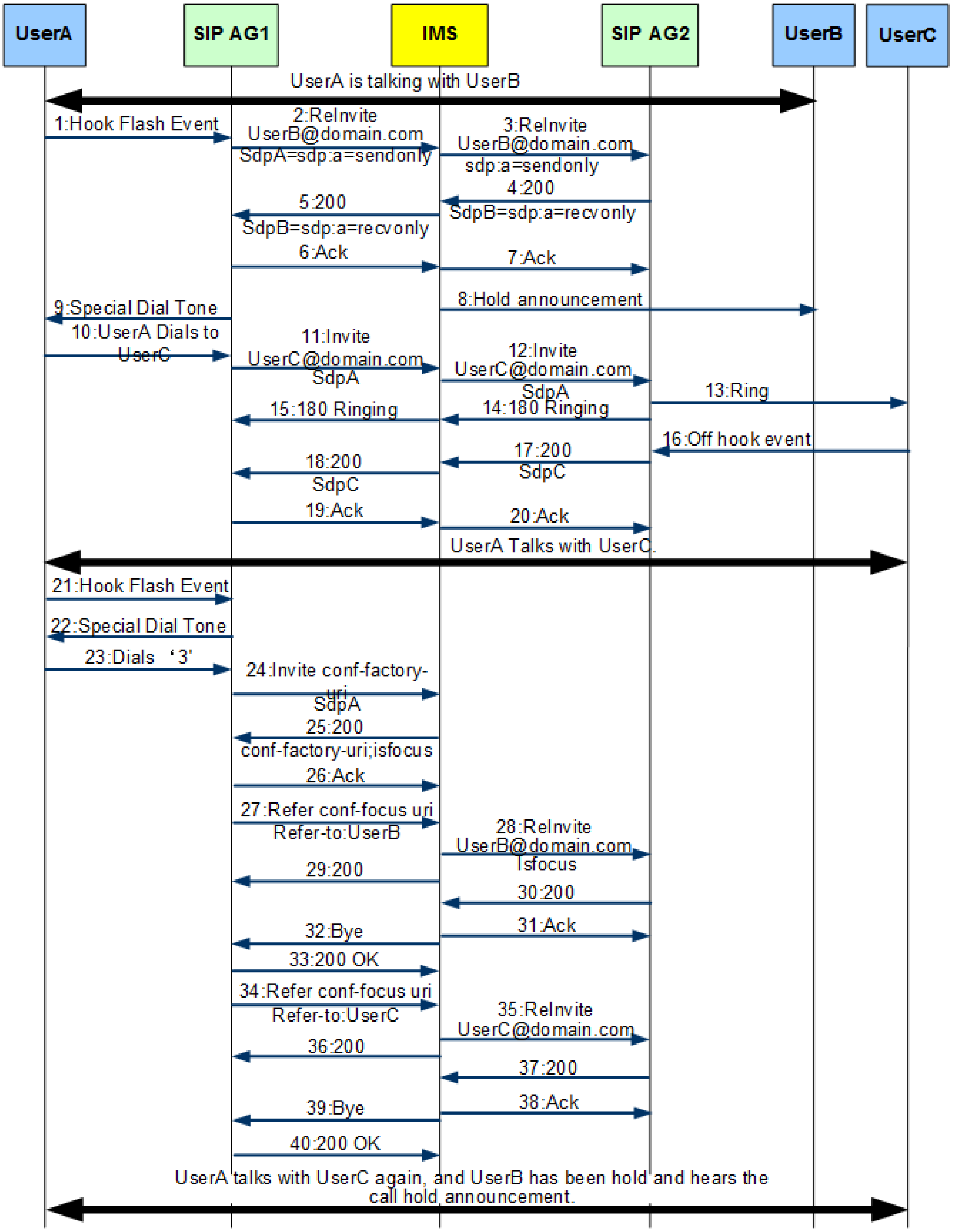
1.6.4.3 Service Flow of Remote Three Party Conference Call

In the following service flows of remote three party conference call (3PTY) for POTS users or ISDN users, user A is the one who subscribes to the service, and users B and C are common users.

Remote 3PTY for POTS Users

Figure 1-56 shows how remote 3PTY for POTS users is implemented.

Figure 1-56 Service flow of remote 3PTY for POTS users


UserA
SIP AG1
IMS
SIP AG2
UserB
UserC
UserA is talking with UserB
1:Hook Flash Event
2:ReInvite UserB@domain.com SdpA=sdp:a=sendonly
3:ReInvite UserB@domain.com sdp:a=sendonly
4:200 SdpB=sdp:a=recvonly
5:200 SdpB=sdp:a=recvonly
6:Ack
7:Ack
8:Hold announcement
9:Special Dial Tone
10:UserA Dials to UserC
11:Invite UserC@domain.com SdpA
12:Invite UserC@domain.com SdpA
13:Ring
14:180 Ringing
15:180 Ringing
16:Off hook event
17:200 SdpC
18:200 SdpC
19:Ack
20:Ack
UserA Talks with UserC
21:Hook Flash Event
22:Special Dial Tone
23:Dials '3'
24:Invite conf-factory-uri SdpA
25:200 conf-factory-uri;isfocus
26:Ack
27:Refer conf-focus uri Refer-to:UserB
28:ReInvite UserB@domain.com Isfocus
29:200
30:200
31:Ack
32:Bye
33:200 OK
34:Refer conf-focus uri Refer-to:UserC
35:ReInvite UserC@domain.com
36:200
37:200
38:Ack
39:Bye
40:200 OK
UserA talks with UserC again, and UserB has been hold and hears the call hold announcement.

The service flow of remote 3PTY for POTS users is as follows:

1. When user A talks with user B, user A presses hookflash.

2. AG1 detects the hookflash pressing event, converts the event into a ReInvite message, and sends the message to user B.
3. After the IMS receives the ReInvite message, it applies for announcement playing resources and sends a ReInvite message to user B. After AG2 receives a ReInvite message, AG2 modifies the media attribute of user B so that user B can receive but cannot send media streams.
4. AG2 replies with a 200 OK response carrying an SDP packet, indicating that user B only receives media streams.
5. The IMS sends a 200 OK response, and AG1 receives the 200 OK message.
6. AG1 sends an Ack message to user B.
7. The IMS sends an Ack message to user B, and AG2 receives the Ack message.
8. User B is successfully placed on hold and hears a call hold tone.
9. User A hears a special dial tone.
10. A dials user C's number.
11. AG1 sends an Invite message to user C after detecting the dialing event.

NOTE

In steps 11 to 19, AG1 initiate a basic call to user C until user C picks up the phone and user A starts to talk with user C. The flow is the same as a basic call flow.

12. After the IMS receives the Invite message, it sends the Invite message to user C. AG2 receives the Invite message.
13. AG2 alerts user C.
14. AG2 sends a 180 response to AG1.
15. The IMS sends the 180 response sent by AG2.
16. User C picks up the phone after hearing a ringing tone.
17. AG2 sends a 200 OK message.
18. The IMS sends the 200 OK message sent by AG2.
19. AG1 sends an Ack message to user C.
20. The IMS sends an Ack message to user C, and AG2 receives the Ack message. Then a call is set up between users A and C.
21. User A presses hookflash again.
22. After pressing hookflash, user A hears a special dial tone.
23. User A dials service code "3" in an attempt to participate in a three-party conference.
24. After AG1 detects the hookflash event and service code "3", AG1 sends an Invite message to the IMS. The message carries **Request-URI** information (indicating the conference factory URI) and applies for a three-party call.
25. After receiving the Invite message, the IMS applies for resources to create a conference. After the conference is successfully created, user A returns the 200 OK message.
26. After AG1 receives the 200 OK message, AG1 sends an Ack message to the IMS.
27. AG1 sends a Refer message to the IMS, inviting user B to join the conference. In the Refer message, **Request-URI** indicates the conference focus URI, and **Refer-To** is user B.
28. The IMS receives the Refer message for inviting user B to the conference and finds that user A is talking with user B. Therefore, the IMS sends a ReInvite message to user B.
29. AG1 receives a 200 OK message from the IMS.
30. After AG2 receives the ReInvite message, AG2 redirects user B to the created conference and sends a 200 message to the IMS.

31. The IMS sends an Ack message to user B, and AG2 receives the Ack message.
32. The IMS sends a Bye message to user A to release the session between users A and B.
33. After AG1 releases the session, AG1 returns a 200 OK response to the IMS.
34. AG1 sends a Refer message to the IMS, inviting user C to join the conference. In the Refer message, **Request-URI** indicates the conference focus URI, and **Refer-To** is user C.
35. The IMS receives the Refer message for inviting user C to the conference and finds that user A is talking with user C. Therefore, the IMS sends a ReInvite message to user C.
36. AG1 receives a 200 OK message from the IMS.
37. After AG2 receives the ReInvite message, AG2 redirects user C to the created conference and sends a 200 message to the IMS.
38. The IMS sends an Ack message to C.
39. The IMS sends a Bye message to user A to release the session between users A and C.
40. After AG1 releases the session, AG1 returns a 200 OK response to the IMS. Then all the three parties (A, B, and C) are in the conference.

NOTE

The service flow for ISDN users is the same as that for POTS users.

Standards Compliance

EN300 359

EN301 134

EN301 065

ETSI TS 300 359

ETSI TS 183 036

ETSI TS 124 642

1.6.5 Conferencing

The conference service is a caller-side service. The conference service allows a user to make a multi-connection call. That is, multiple users (participants) can participate in the same session at the same time.

1.6.5.1 Conferencing

This topic describes the definition and benefits of the conferencing service.

Definition

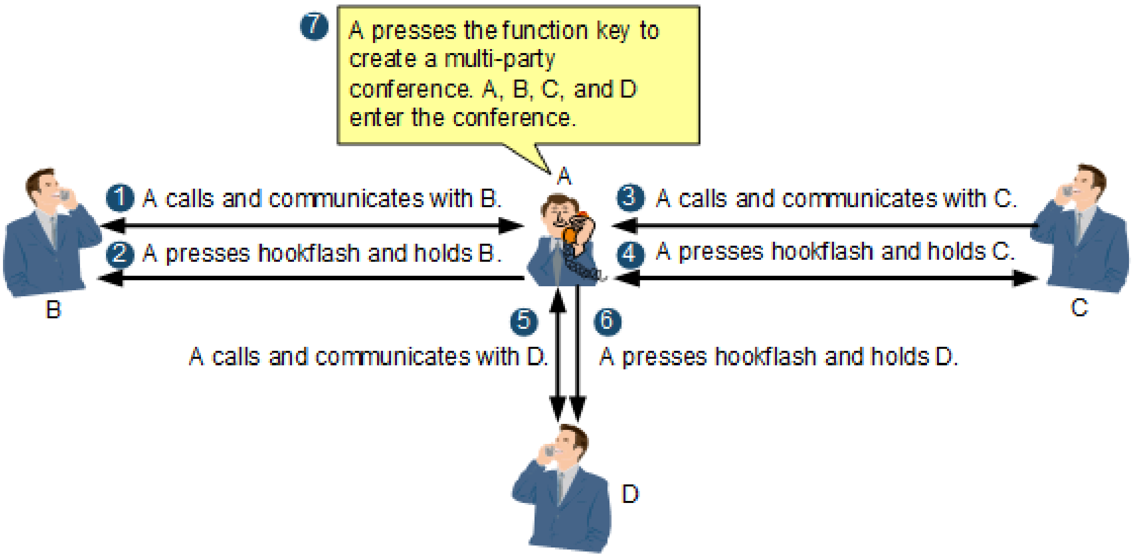
The conferencing service is a service on the originating side. It provides the multi-connection call capability for a user. That is, multiple users (participants) can participate in the same session at the same time. If a user has enabled and activated the service, the user can apply for the resource to create a conference and add or delete the conference participant. The remote user can leave the conference.

NOTE

The difference between the conference service and the three party service is that the conference can be performed only by the Multimedia Resource Function Controller (MRFC), and the number of conference participants can be more than 3.

User A has the conferencing service permission. Users B, C, and D are common users. User A has created a conferencing service with users B, C, and D. Figure 1-57 shows the conferencing service flow.

Figure 1-57 Conferencing service flow


7 A presses the function key to create a multi-party conference. A, B, C, and D enter the conference.
A
1 A calls and communicates with B.
2 A presses hookflash and holds B.
3 A calls and communicates with C.
4 A presses hookflash and holds C.
B
C
5 A calls and communicates with D.
6 A presses hookflash and holds D.
D

1. User A calls user B, and the call between users A and B is set up.
2. User A presses hookflash. User B hears a call hold tone.
3. User A calls user C and communicates with user C.
4. User A presses hookflash to hold user C, and user C hears a call hold tone.
5. User A calls user and communicates with user D.
6. User A presses hookflash to hold user D, and user D hears a call hold tone.
7. User A presses the function key to set up a conference call. Users A, B, C, and D enter a conference call.

Benefit

Beneficiary	Benefits
Carrier	The conferencing service brings added value and provides multi-party session services for users, improving user satisfaction.
User	A user can initiate a conference call at any time and can discuss with multiple users at the same time, which facilitates communication between multiple users and reduces the communication and information transmission costs between multiple users.

Standards Compliance

- EN300 185
- ETSI TS 183 005

- ETSI TS 183 036
- ETSI TS 183 043
- ETSI TS 124 505
- ETSI 300 102-1
- 3GPP TS 24147
- 3GPP TS 24605
- ITU-T Q.931

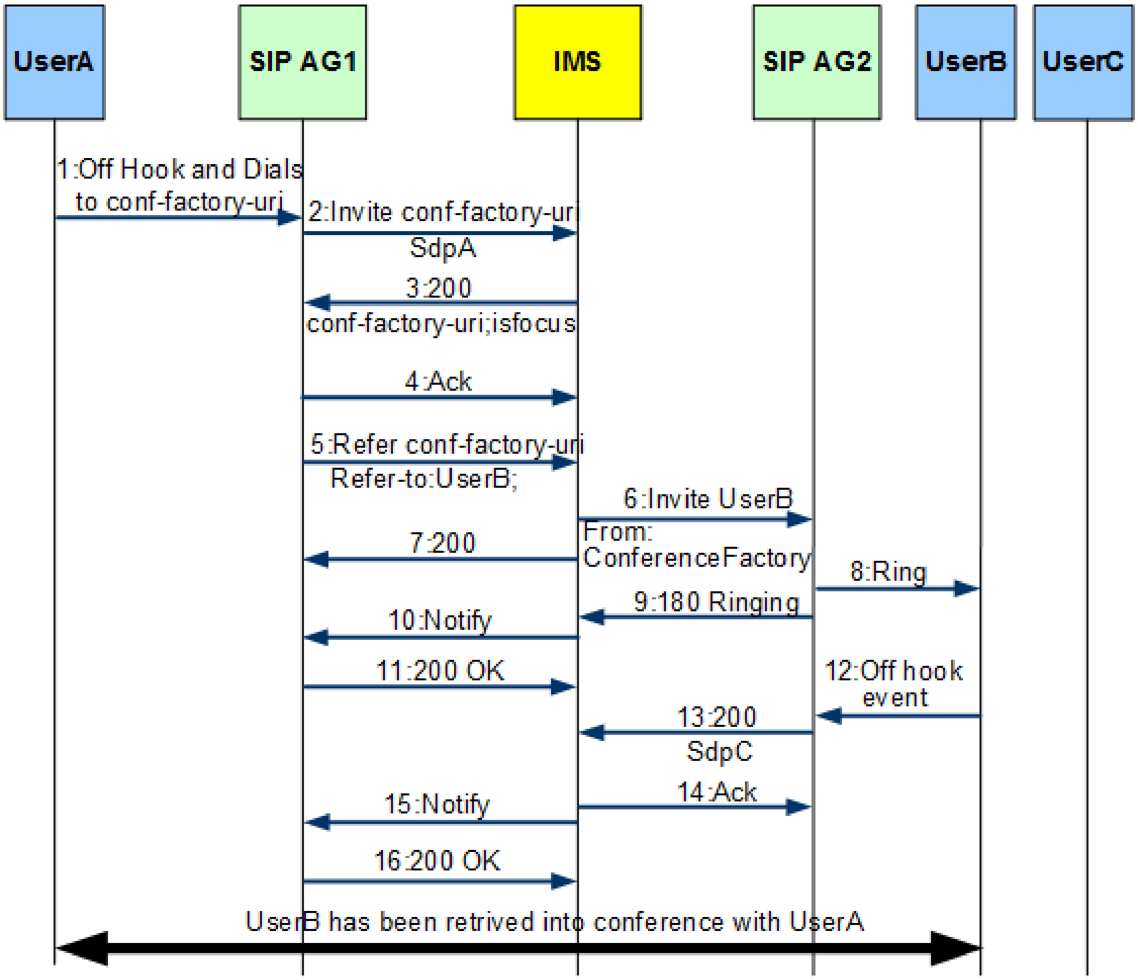
1.6.5.2 Conferencing Service Flow

This topic takes users A, B, and C as an example to describe the POTS and ISDN conferencing service flows.

POTS Conferencing Service Flow

Figure 1-58 shows the POTS conferencing service flow.

Figure 1-58 POTS conferencing service flow


UserA
SIP AG1
IMS
SIP AG2
UserB
UserC
1:Off Hook and Dials to conf-factory-uri
2:Invite conf-factory-uri SdpA
3:200 conf-factory-uri;isfocus
4:Ack
5:Refer conf-factory-uri Refer-to:UserB;
6:Invite UserB From: ConferenceFactory
7:200
8:Ring
9:180 Ringing
10:Notify
11:200 OK
12:Off hook event
13:200 SdpC
14:Ack
15:Notify
16:200 OK
UserB has been retrived into conference with UserA

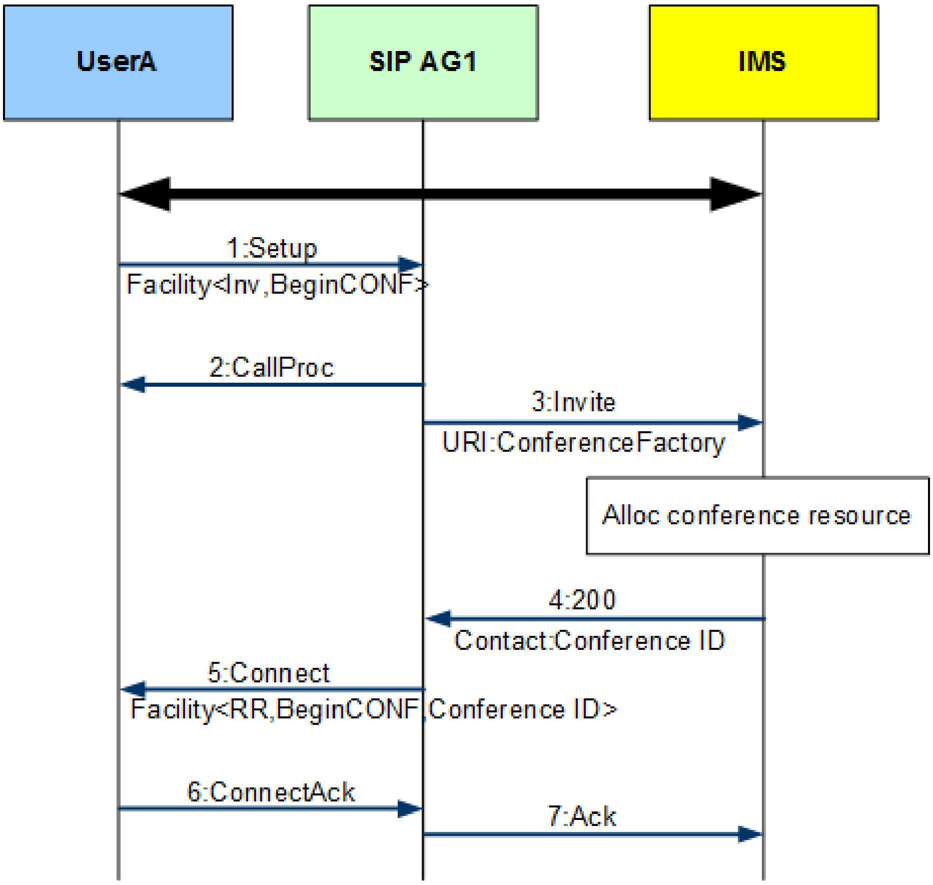
The POTS conferencing service flow is as follows:

1. User A picks up the phone and makes a conference call.
2. AG1 sends an Invite message to the IMS. The Request-URI in the Invite message is a conference factory URI).
3. After receiving the Invite message, the IMS applies for resources to create a conference. After the conference is successfully created, user A returns the 200 OK message.
4. After receiving the 200 OK message, AG1 sends an Ack message to the IMS.
5. AG1 sends a Refer message to the IMS, inviting user B to join the conference. The Request-URI in the Refer message is the conference URI, and the Refer-To is user B.
6. After receiving the Refer message, the IMS sends an Invite message to user B and calls user B.
7. The IMS returns the 200 OK Refer message to user A.
8. After receiving the Invite message from user B, AG2 sends the ring tone to user B.
9. AG2 sends back a 180 ringing message to the IMS.
10. The IMS sends a Notify message to notify user A.
11. AG1 sends a 200 OK response to the Notify message from the IMS.
12. User B picks up the phone.
13. AG2 returns a 200 OK response to the Invite message from the IMS.
14. The IMS sends an Ack message to user B.
15. The IMS sends a Notify message to notify user A.
16. AG1 sends a 200 OK response to the Notify message from the IMS. Then, both users A and B join the conference. The service flow of joining a conference by other users (such as users C and D) is similar to steps 5 to 16.

ISDN User Conferencing Service Flow (Adding a Service Party to a Conference in Idle Mode)

Figure 1-59 shows the service flow of an ISDN user (service party) joining a conference in idle mode.

Figure 1-59 Service flow of an ISDN user (service party) joining a conference in idle mode



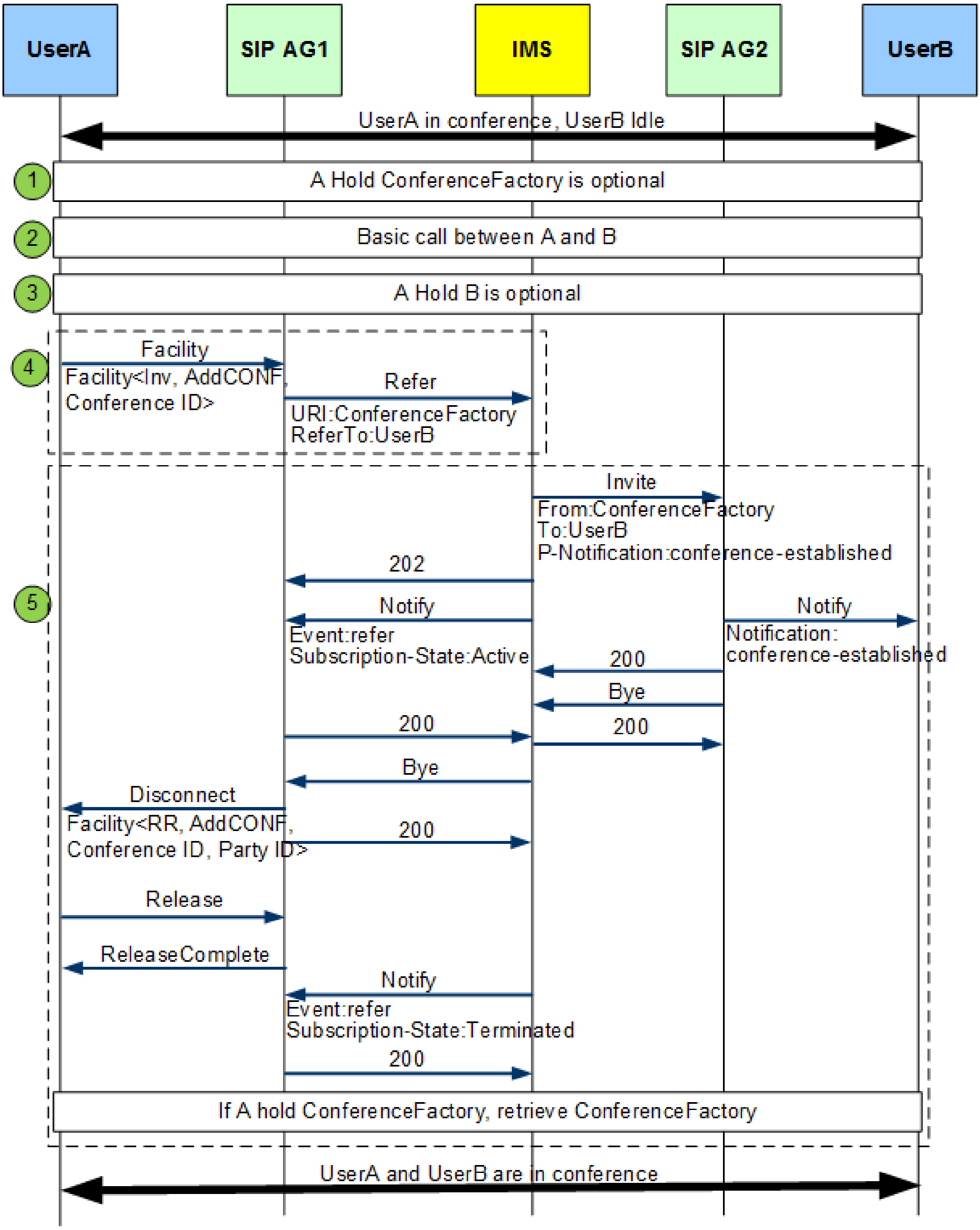
User A is the conferencing service party. The service flow of an ISDN user (service party) joining a conference in idle mode is as follows:

1. User A sends a SETUP message carrying FacilityIE to AG1, asking to set up a conference.
2. AG1 sends a dial tone to user A. User A dials a number. AG1 receives the number and matches the digitmap successfully. The entire flow is the same as a basic call flow.
3. AG1 applies to the IMS conference factory for creating a conference. The IMS allocates conference resources.
4. The IMS returns a 200 OK response carrying the conference ID to AG1.
5. AG1 sends a Connect message carrying Facility to user A. The Connect message carries the conference ID, indicating that A joins the conference successfully.
6. User A replies AG1 with a ConnectAck message.
7. AG1 sends an Ack message to the IMS. Then, user A joins the conference in the idle state.

ISDN User Conferencing Service Flow (Adding a Party to a Conference)

Figure 1-60 shows the service flow of an ISDN user adding a party to a conference.

Figure 1-60 Service flow of adding the other party to a conference


UserA
SIP AG1
IMS
SIP AG2
UserB
UserA in conference, UserB Idle
1
A Hold ConferenceFactory is optional
2
Basic call between A and B
3
A Hold B is optional
4
Facility
Facility<Inv, AddCONF, Conference ID>
Refer
URI:ConferenceFactory
ReferTo:UserB
Invite
From:ConferenceFactory
To:UserB
P-Notification:conference-established
202
5
Notify
Event:refer
Subscription-State:Active
Notify
Notification:
conference-established
200
Bye
200
200
Bye
Disconnect
Facility<RR, AddCONF, Conference ID, Party ID>
200
Release
ReleaseComplete
Notify
Event:refer
Subscription-State:Terminated
200
If A hold ConferenceFactory, retrieve ConferenceFactory
UserA and UserB are in conference

User A is the conferencing service party. The service flow of adding the other party to a conference is as follows:

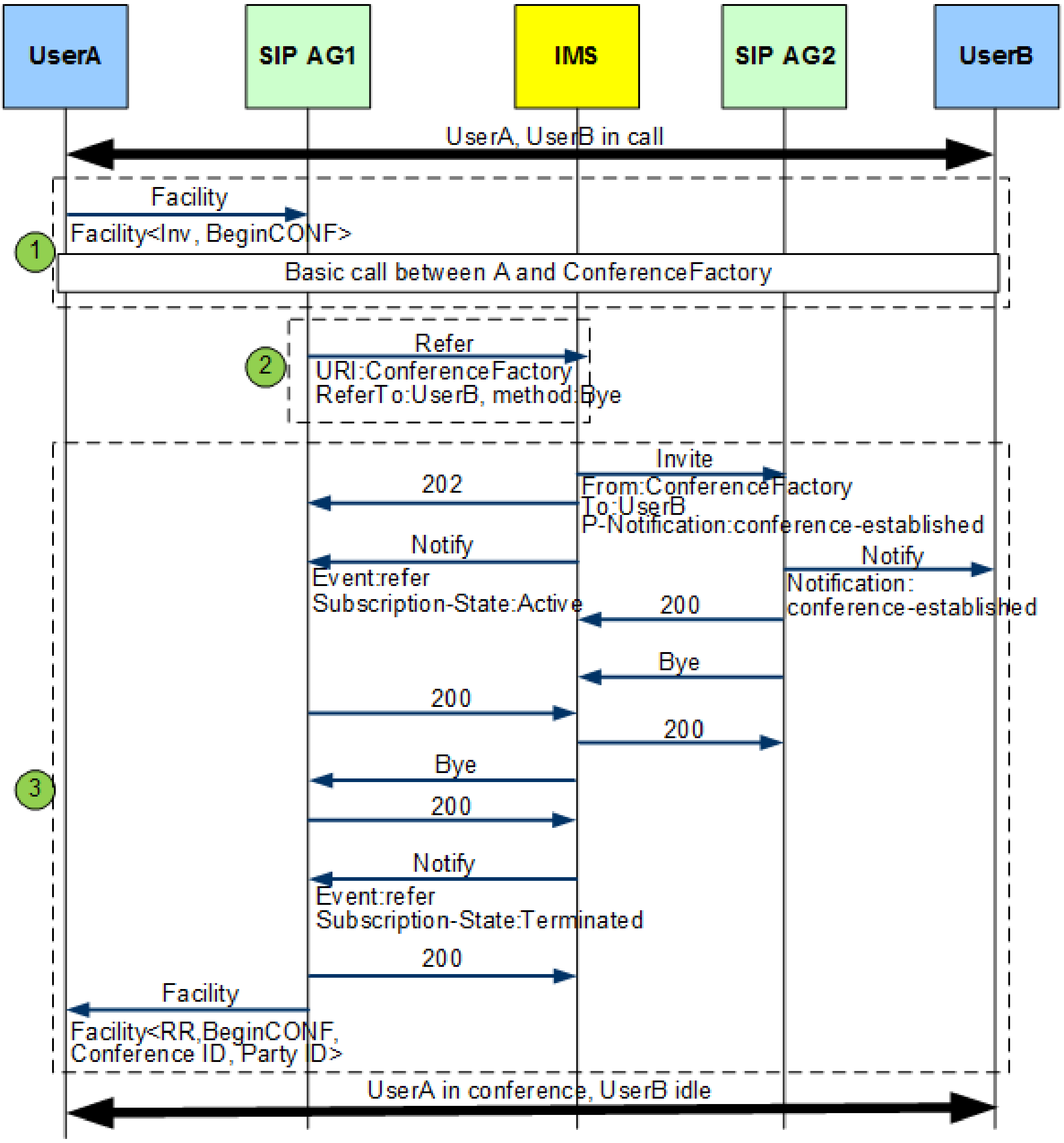
1. During a conference, user A can hold a call with the conference factory before initiating a call with user B.

2. User A (service party) and user B (non-service party) set up a basic call. The call can also be set up by user B.
3. User A can hold the call with user B before adding user B to the conference.
4. User A requests to add user B to the conference through a terminal operation. AG1 sends a Refer message to the IMS, asking the IMS to add user B to the conference.
5. After receiving the Refer request, the IMS establishes a new call between the conference factory and user B, and releases the call between users A and B.

ISDN User Conferencing Service Flow (Service Party Joining a Conference in the Running State)

Figure 1-61 shows the service flow of an ISDN user joining a conference in the running state.

Figure 1-61 Service flow of an ISDN user joining a conference in the running state



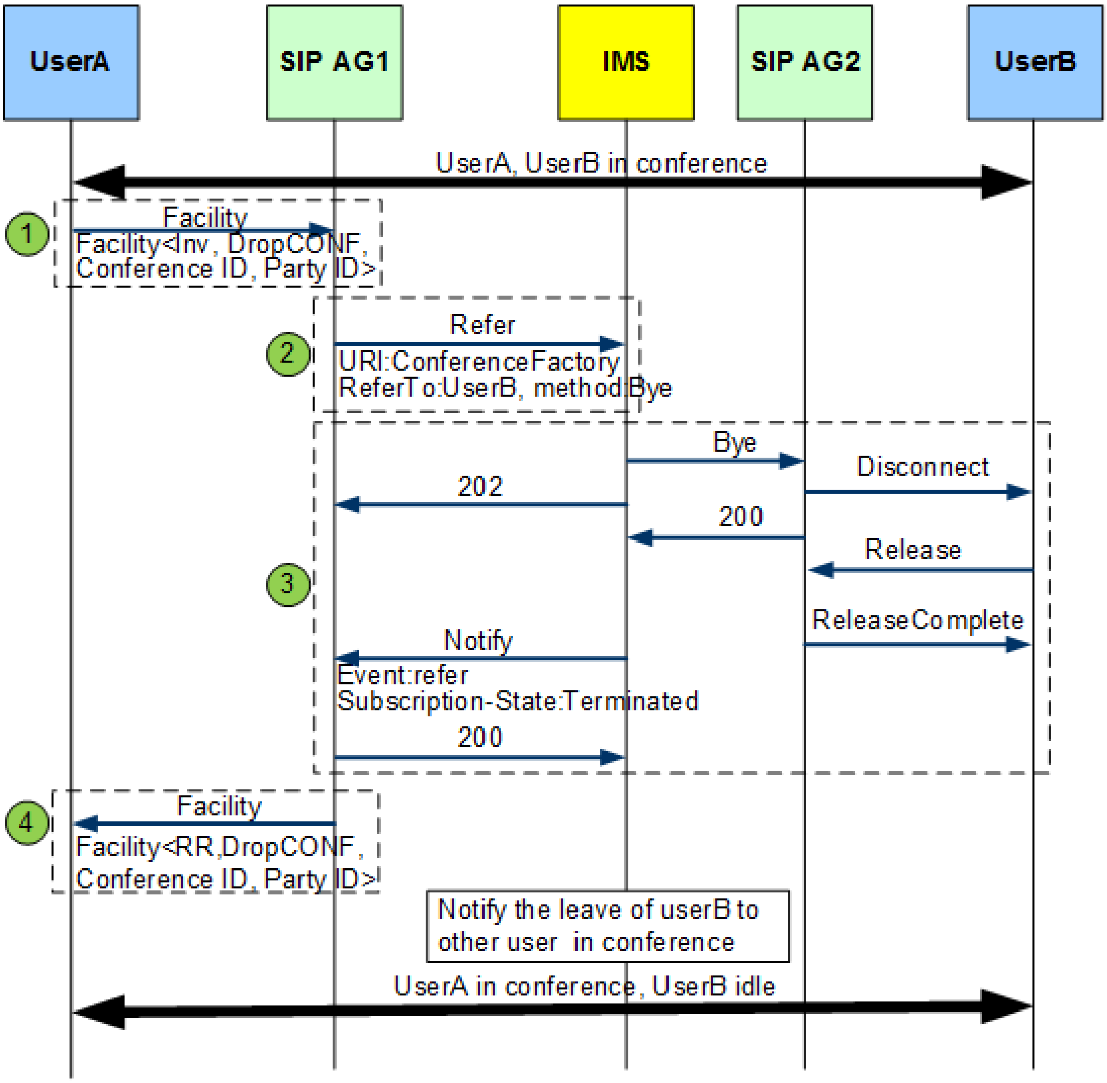
User A is the conferencing service party. Before the conference, users A and B are in a basic call. The service flow of the ISDN user in the running state is as follows:

1. User A sends a Facility message to AG1, asking to set up a conference. AG1 sends an Invite request to the IMS and conference factory to set up a conference. The IMS accepts the request. The service flow is the same as that in the idle state. For details, see ISDN User Conferencing Service Flow (Adding a Service Party to a Conference in Idle Mode).
2. AG1 sends a Refer message to the IMS, inviting user B to join the conference.
3. After receiving the Refer request, the IMS establishes a new call between the conference factory and user B and releases the call between users A and B.

ISDN User Conferencing Service Flow (Disconnecting One Party in a Conference)

Figure 1-62 shows the service flow of disconnecting a party from an ISDN user conference.

Figure 1-62 Service flow of disconnecting a party from an ISDN user conference



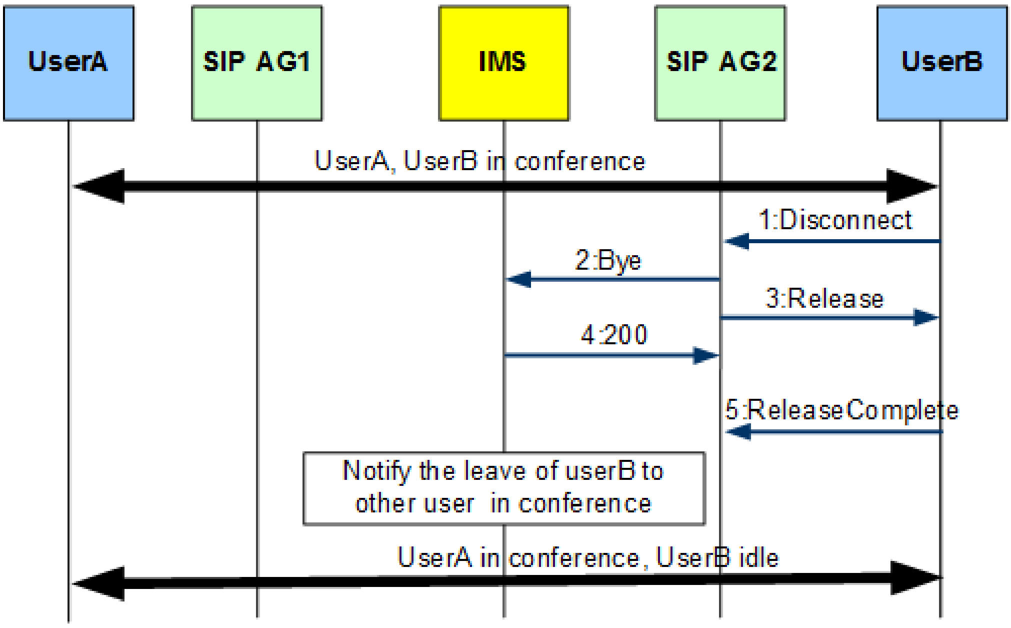
The service flow of disconnecting a party from an ISDN user conference is as follows:

1. User A requests to disconnect user B from the conference.
2. AG1 sends a Refer message to the IMS, asking the IMS to disconnect user B from the conference.
3. After receiving the Refer request, the IMS releases the call between the conference factory and user B, and notifies AG1 of user A.
4. AG1 sends a Facility response to notify user A of the disconnection result.

ISDN User Conferencing Service Flow (Non-Service Party Quitting a Conference)

Figure 1-63 shows the service flow of an ISDN user (non-service party) exiting a conference.

Figure 1-63 Service flow of an ISDN user (non-service party) exiting a conference

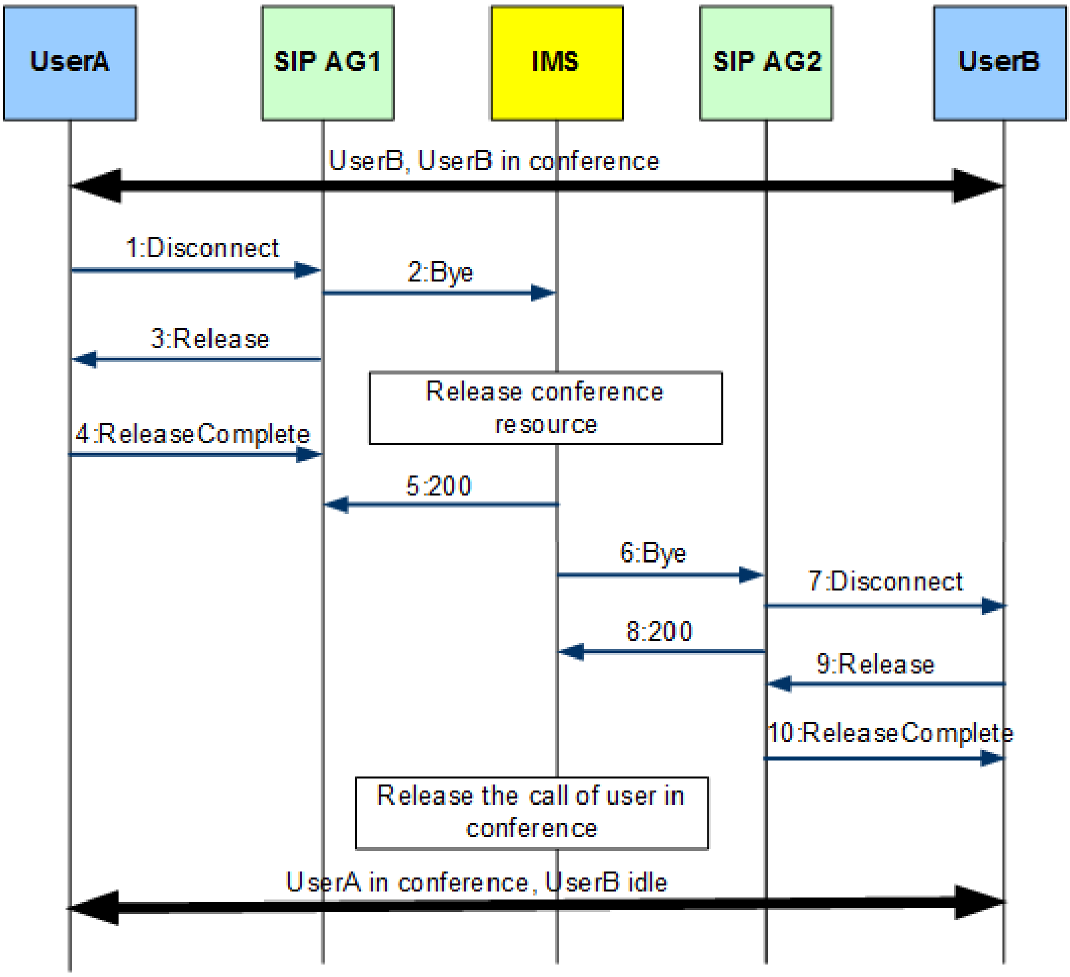

UserA
SIP AG1
IMS
SIP AG2
UserB
UserA, UserB in conference
1:Disconnect
2:Bye
3:Release
4:200
5:ReleaseComplete
Notify the leave of userB to other user in conference
UserA in conference, UserB idle

In the preceding figure, steps 1 to 5 indicate that user B releases the call with the conference factory. Then, the IMS notifies conference participants that participant B leaves the conference.

ISDN User Conferencing Service Flow (Releasing a Conference by the Service Party)

Figure 1-64 shows the service flow of an ISDN user (service party) releasing a conference.

Figure 1-64 Service flow of an ISDN user (service party) releasing a conference



User A is the conferencing service party. The service flow of an ISDN user (service party) releasing a conference is as follows:

- Steps 1 to 6: User A (service party) releases the call with the conference factory.
- Steps 7 to 10: The IMS releases the conference resources and releases the call between the conference factory and other participants.

1.6.6 Explicit Communication Transfer

This service allows user A who is communicating with user B to transfer the call to user C so that a call is set up between user B and user C.

1.6.6.1 Explicit Communication Transfer

This topic describes the definition and principle of the explicit communication transfer (ECT) service.

Definition

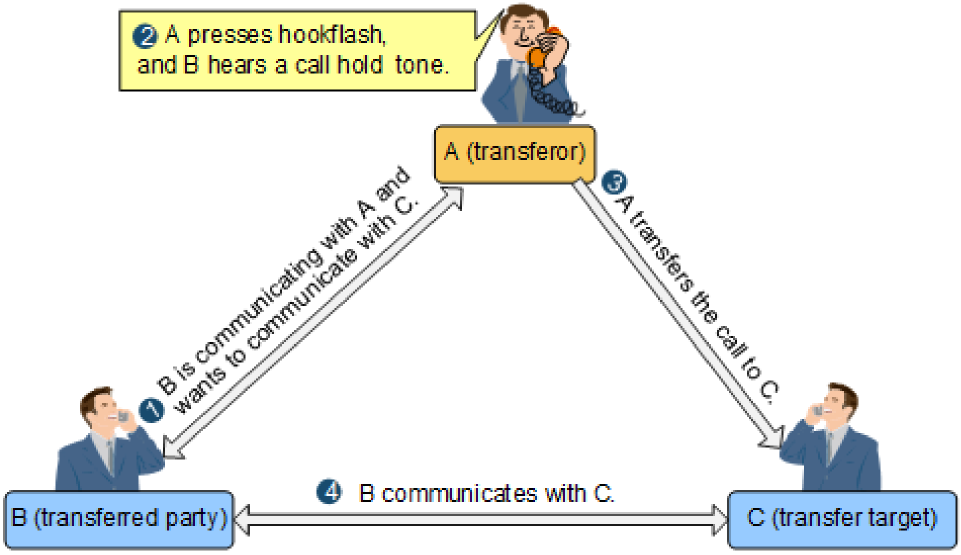
The ECT service enables user A who is in a conversation with user B to transfer the call to user C so that the call between user B and user C is set up.

The ECT service for POTS users includes 2 scenarios: off-hook ECT and on-hook ECT. The off-hook ECT service can be consultative or blind, and the on-hook ECT service can be during conversation or during ringing.

The ECT service for ISDN users includes 2 scenarios: implicit transfer and explicit transfer.

User A has subscribed to the ECT service. User A is the transferor, user B is a transferee, and user C is also a transferee. Figure 1-65 shows the ECT service flow (using an off-hook blind transfer scenario as an example).

Figure 1-65 Principle of the ECT service (off-hook blind transfer)


❷ A presses hookflash, and B hears a call hold tone.
A (transferor)
❶ B is communicating with A and wants to communicate with C.
❸ A transfers the call to C.
❹ B communicates with C.
B (transferred party)
C (transfer target)

1. User A is communicating with user B. User B wants to communicate with user C.
2. User A presses the hook switch. User B is held and hears the call hold tone, and user A hears a dial tone.
3. User A dials user C's number to call user C.
4. User C's phone rings. User A initiates a call transfer and is released. User B hears the ring back tone.
5. User C answers the call. User B communicates with user C.

NOTE

If the call fails to be transferred, user B initiates a renegotiation with user A to resume the call between users A and B.

Benefit

Beneficiary	Benefits
Carrier	The ECT service enhances carriers' competitiveness and provides carriers with new revenue growth potential.
User	The ECT service allows a user to easily transfer a call to another user so that the desired user can answer the call.

Standards Compliance

- ETSI TS 124 529
- ETSI EN 300 102-1
- ETSI EN 300 369-1
- ETSI TS 183 029
- ETSI TS 183 043
- ETSI TS 183 036
- ITU-T Q.931

1.6.6.2 Service Flow of Explicit Communication Transfer

The explicit communication transfer (ECT) service for POTS users includes 2 scenarios: off-hook ECT and on-hook ECT. The off-hook ECT service can be consultative or blind, and the on-hook ECT service can be during conversation or during ringing. The ECT service for ISDN users includes 2 scenarios: implicit transfer and explicit transfer. This topic describes the flows of the preceding ECT services.

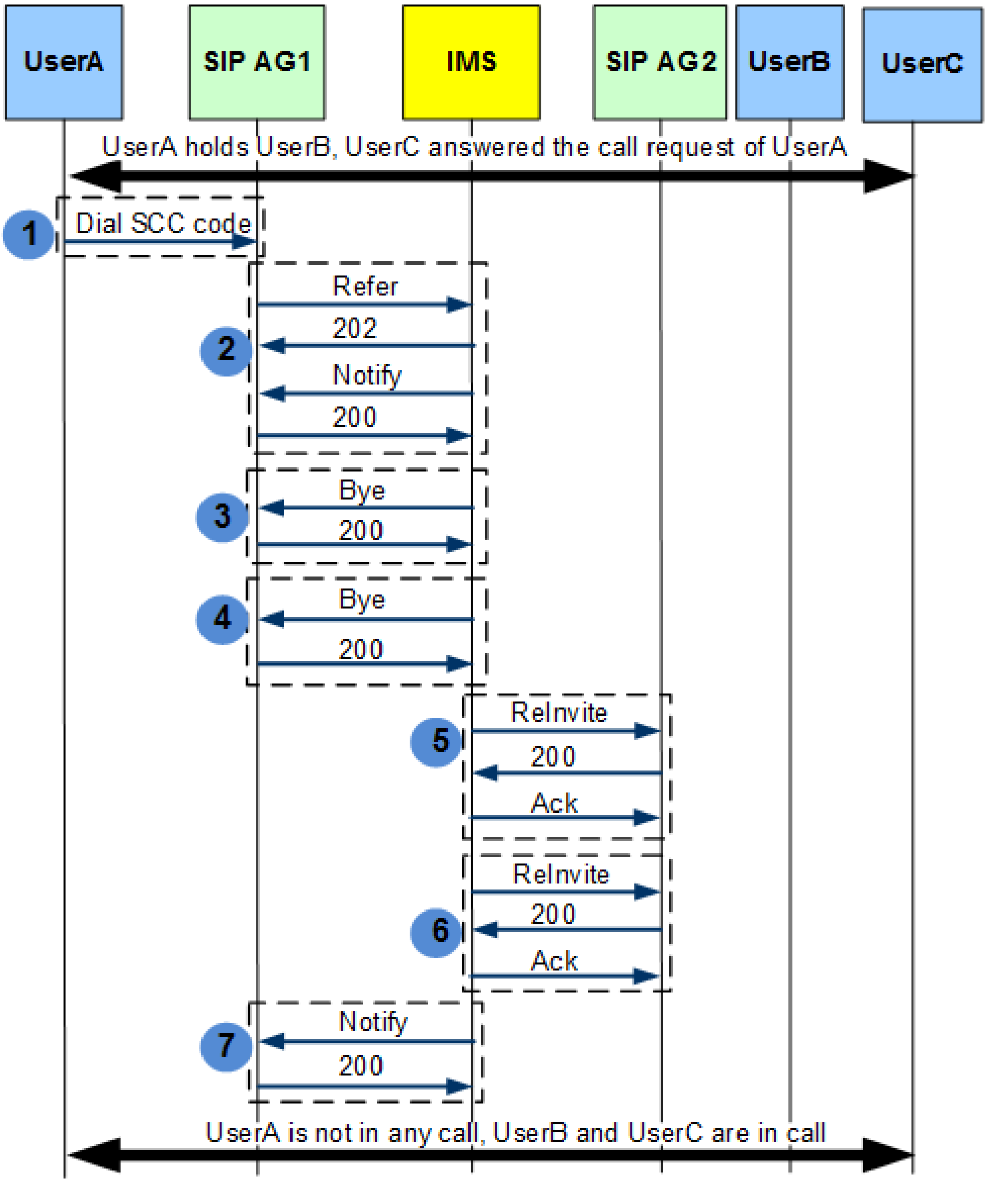
In the following message flows, user A has subscribed to the service and is the transferor, user B is the transferee, and user C is the transfer target. For ease of description, the service flows are based on the following user calls:

- User B sets up a call with user A. After the call is set up, user A places user B on hold.
- User A establishes a call to user C. After the call is set up or the phone of user C rings, user A hangs up and initiates a call transfer.
- User B talks with user C.

Consultative Transfer in Off-Hook State for POTS Users (Off-Hook Consultative Transfer)

Figure 1-66 shows how off-hook consultative transfer is implemented.

Figure 1-66 Service flow of off-hook consultative transfer for POTS users



Transferor A and transferee B enter a conversation through a basic call. Transferor A presses hookflash and enters the call hold state with transferee B through a call hold flow. Transferor A enters a conversation with transfer target C through a basic call. Transferor A presses hookflash and enters the call hold state with transfer target C through a call hold flow. The service flow of off-hook consultative transfer for POTS users is as follows:

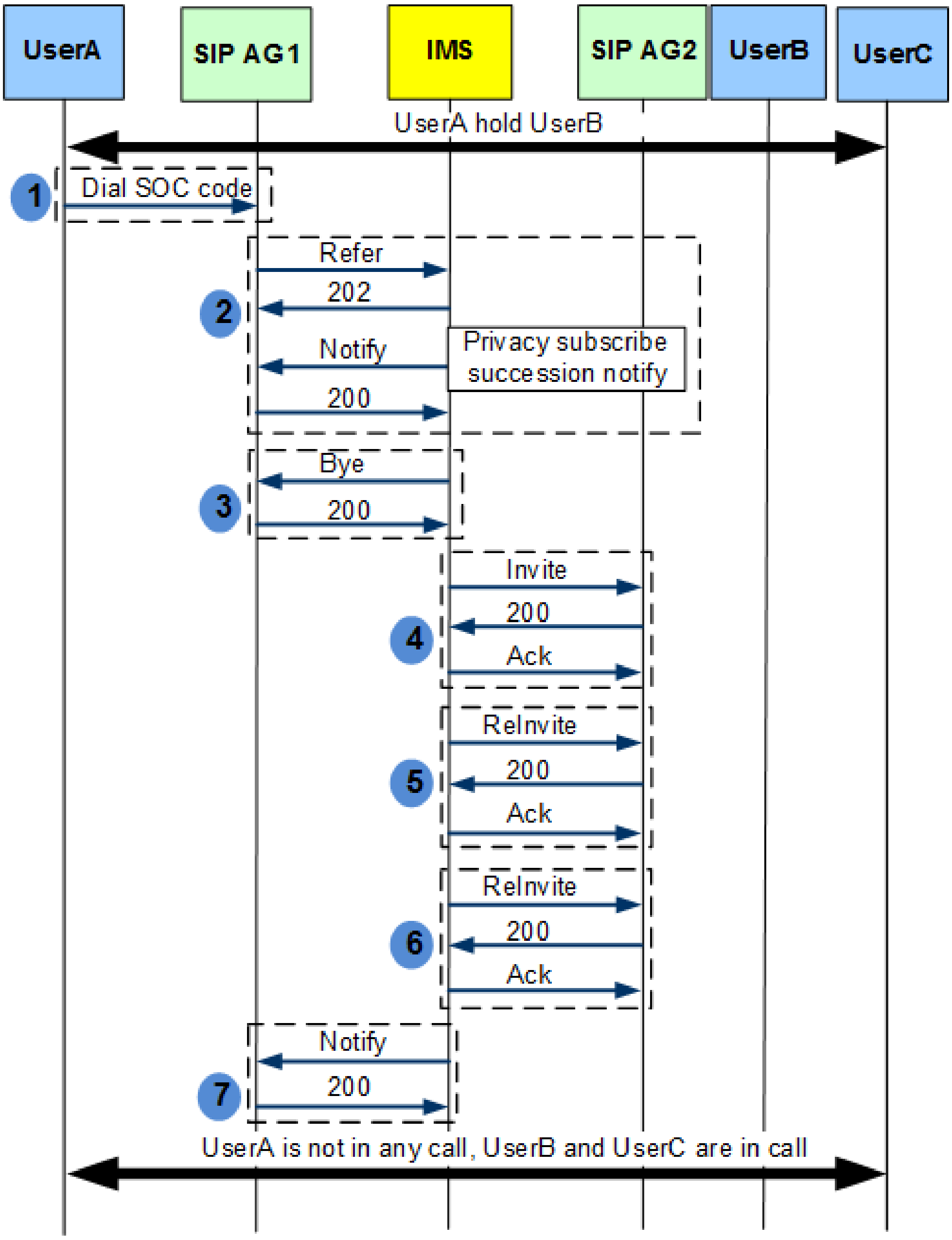
1. User A dials an explicit transfer SCC number.
2. User A sends a Refer message to request the call transfer. The IMS sends a Notify message.
3. The IMS sends a Bye message to release the session between users A and B.
4. The IMS sends a Bye message to release the session between users A and C.
5. The IMS sends a ReInvite message to user B for media renegotiation. User B replies with a 200 message.

6. The IMS sends a ReInvite message to user C for media renegotiation. User C replies with a 200 message. The media renegotiation is successful. Users B and C are engaged in a call.
7. The IMS sends a Notify message, notifying the transferor that the call is successfully transferred. If user A does not hang up, the system plays a short busy tone to notify that the call is successfully transferred.

Blind Transfer in Off-Hook State for POTS Users (Off-Hook Blind Transfer)

Figure 1-67 shows how off-hook blind transfer is implemented.

Figure 1-67 Service flow of off-hook consultative transfer for POTS users



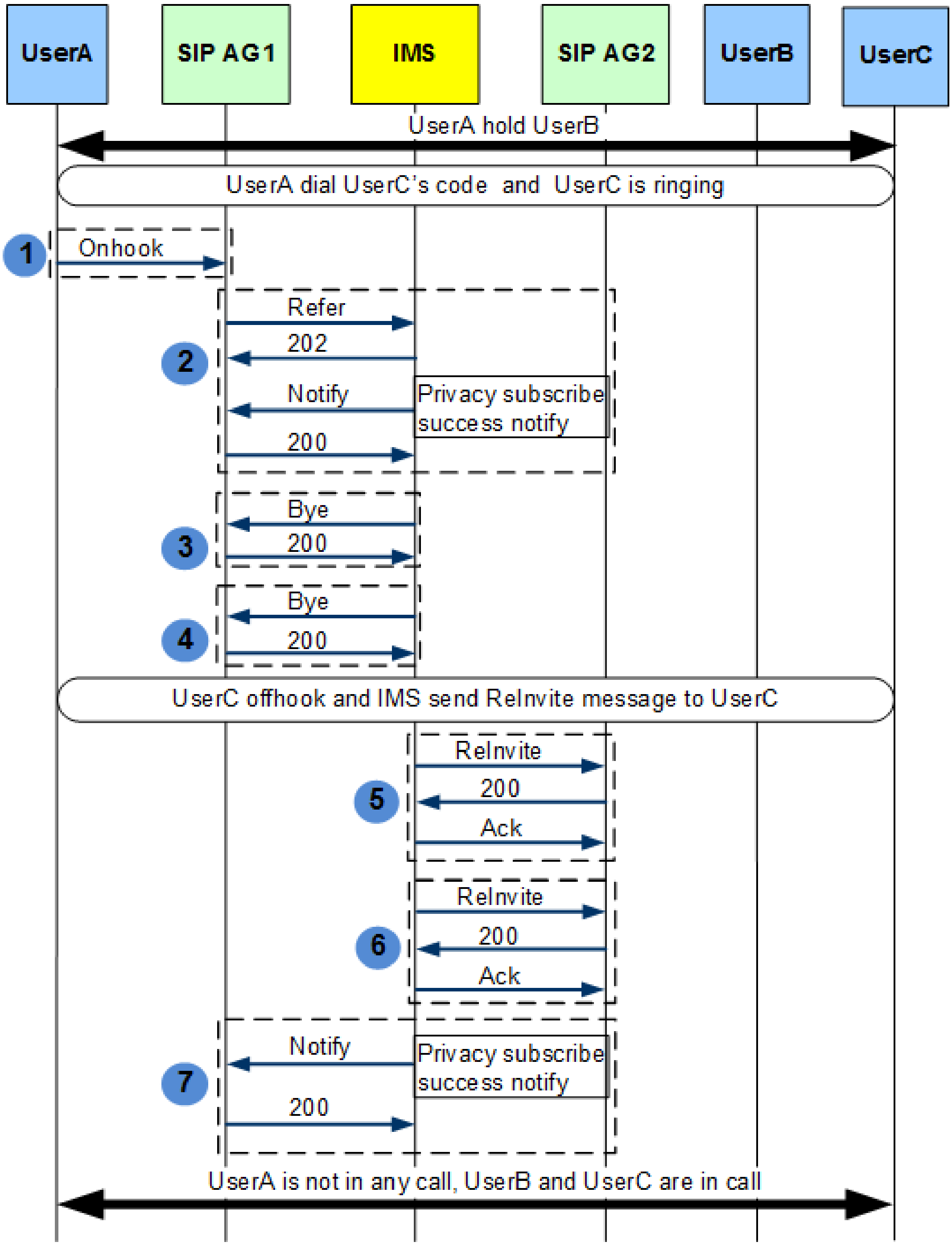
Transferor A and transferee B enter a conversation through a basic call. User A presses hookflash and enters the call hold state with user B through a call hold flow. The service flow of off-hook blind transfer for POTS users is as follows:

1. User A dials a blind transfer SOC number. The number format is as follows: 4*TransferTarget TelNum.
2. User A sends a Refer message to request a blind transfer. The IMS sends a Notify message, notifying A that the subscription is successful.
3. The IMS sends a Bye message to release the session between users A and B. Because the subscription exists, dialog D1 still exists.
4. The IMS sends an Invite message to initiate a new basic call to user C. User C replies with a 200 message.
5. The IMS sends a ReInvite message to user B for media renegotiation. User B replies with a 200 message.
6. The IMS sends a ReInvite message to user C for media renegotiation. User C replies with a 200 message. The media renegotiation is successful. Users B and C are engaged in a call.
7. The IMS sends a Notify message, notifying the transferor that the call is successfully transferred. If user A does not hang up, the system plays a short busy tone to notify that the call is successfully transferred.

On-Hook Transfer During Ringing for POTS Users (On-Hook Consultative Transfer During Ringing)

Figure 1-68 shows how on-hook consultative transfer during ringing is implemented.

Figure 1-68 Service flow of on-hook consultative transfer during ringing for POTS users


UserA
SIP AG1
IMS
SIP AG2
UserB
UserC
UserA hold UserB
UserA dial UserC's code and UserC is ringing
1
Onhook
Refer
202
2
Notify
Privacy subscribe success notify
200
Bye
3
200
Bye
4
200
UserC offhook and IMS send ReInvite message to UserC
ReInvite
5
200
Ack
ReInvite
6
200
Ack
Notify
7
Privacy subscribe success notify
200
UserA is not in any call, UserB and UserC are in call

Users A and B enter a call through a basic call. User A presses hookflash and enters the call hold state with user B through a call hold flow. User A dials user C's number to initiate a basic call to user C. User A receives a 180 response and enters the ring back tone phase. The service flow is as follows:

1. User A hangs up.
2. User A sends a Refer message to request a consultative transfer. The IMS sends a Notify message to notify user A.

3. The IMS sends a Bye message to release the session between users A and B.
4. The IMS sends a Bye message to release the session between users A and C.
5. After user C picks up the phone, the IMS performs media renegotiation and sends a ReInvite message to user C.
6. The IMS sends a ReInvite message to user B for media renegotiation. Then the media renegotiation is successful, and users B and C enter a conversation.
7. The IMS sends a Notify message, notifying user A that the call is successfully transferred.

On-Hook Transfer During Conversation for POTS Users (On-Hook Consultative Transfer During Conversation)

The service flow of on-hook consultative transfer during conversation for POTS users is as follows:

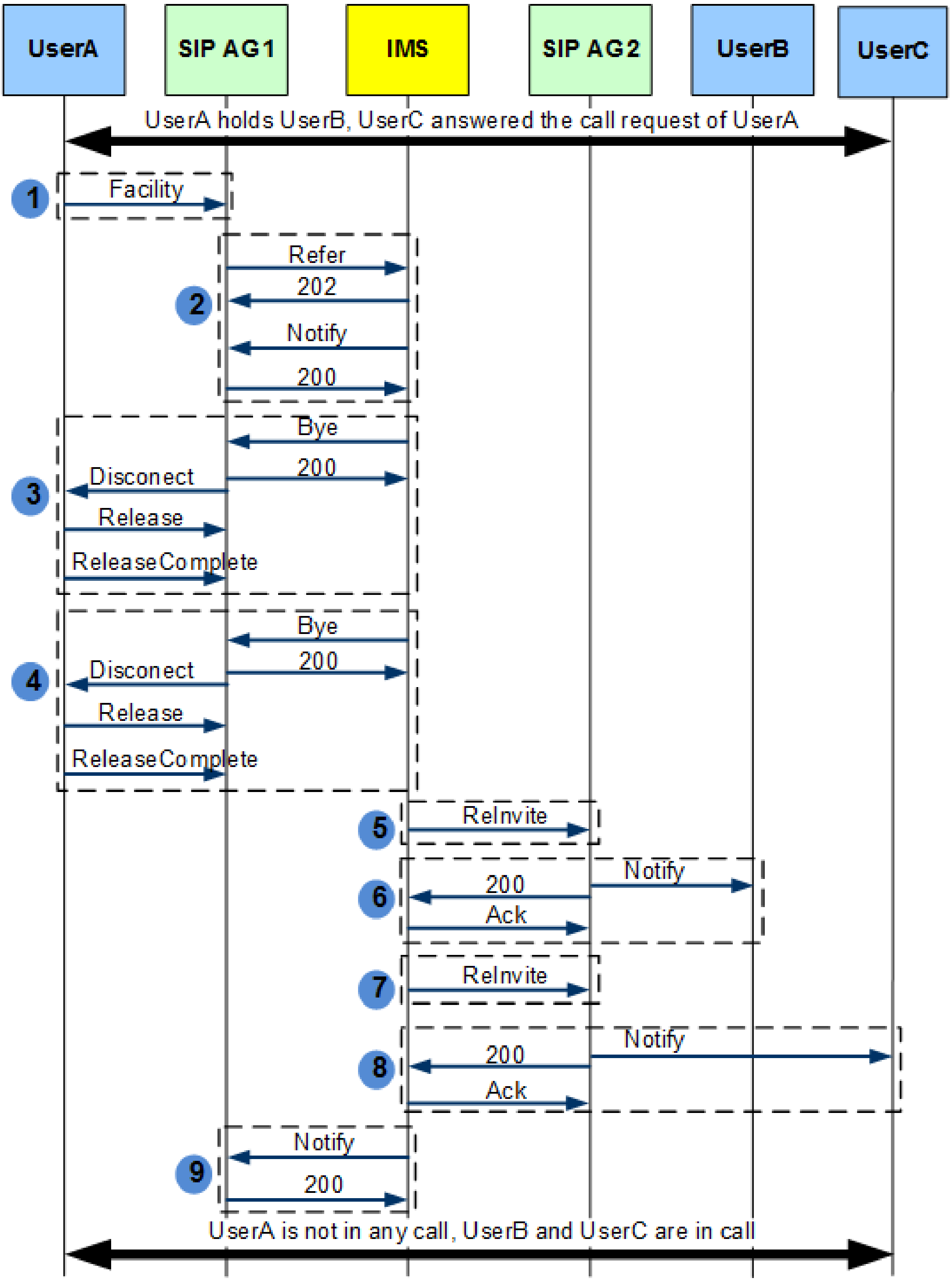
Users A and B enter a call through a basic call. User A presses hookflash and enters the call hold state with user B through a call hold flow. User A dials user C's number and enters a call with user C.

The subsequent steps are the same as those described in On-Hook Transfer During Ringing for POTS Users (On-Hook Consultative Transfer During Ringing).

Implicit Transfer for ISDN Users (When User C Answers a Call)

If user C answers a call and then user A initiates a call transfer, the implicit transfer service is implemented as shown in Figure 1-69.

Figure 1-69 Service flow of implicit communication transfer for ISDN users (when user C answers a call)



After user C answers a call, the implicit transfer service is implemented as follows:

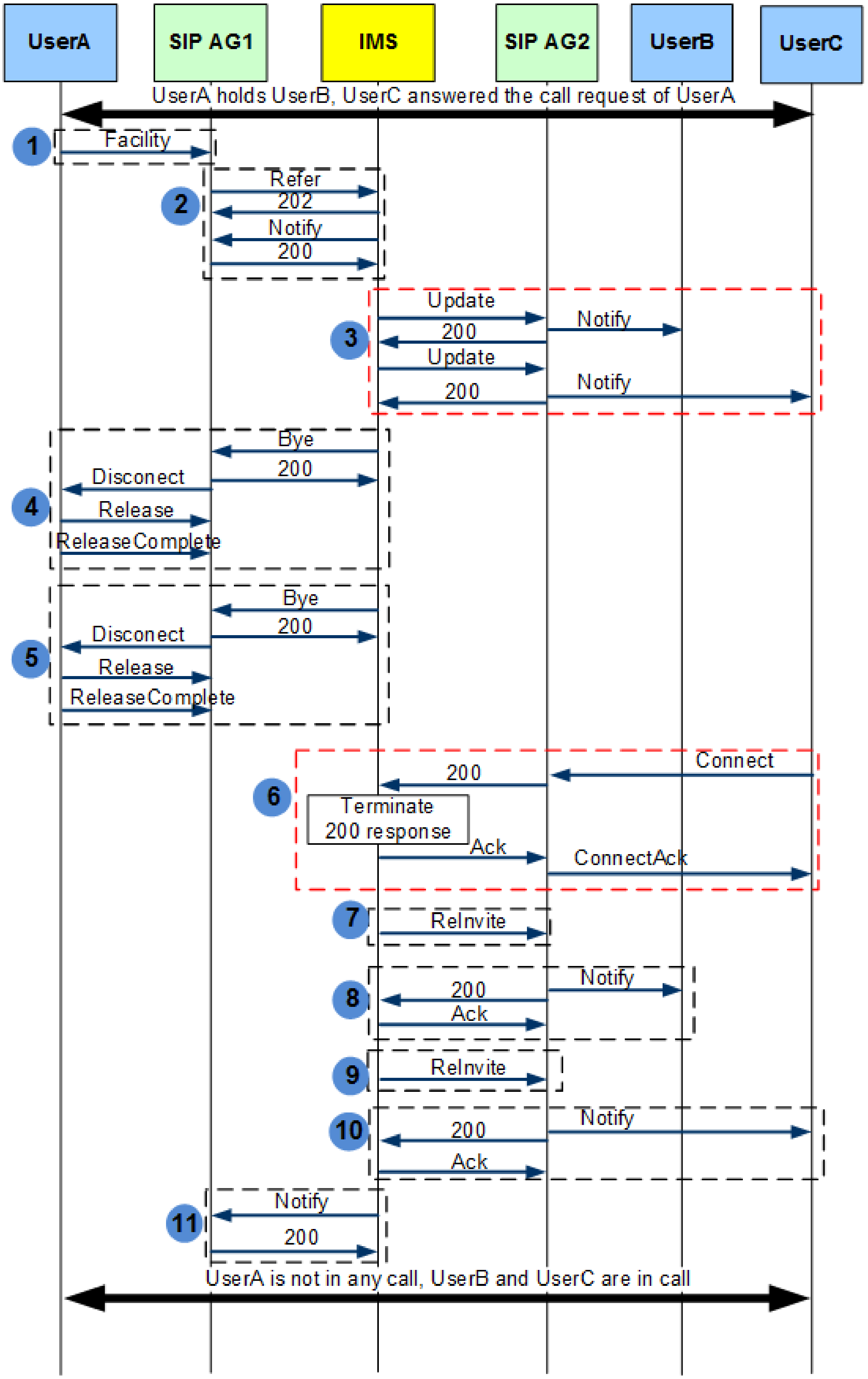
1. During a call between users A and B, when user C answers the call, user A sends a Facility message to AG1, instructing AG1 to transfer the call.
2. AG1 sends a Refer message to the IMS for a call transfer.

3. The IMS releases the session between users A and B.
4. The IMS releases the session between users A and C.
5. The IMS sends a ReInvite message to user B, which carries SDP data to instruct user B to establish a media stream between users B and C and carries P-NOTIFICATION header field data to instruct a call transfer.
6. After AG2 (for user B) receives the ReInvite message for a call transfer, AG2 converts the message into a Notify message to notify user B.
7. The IMS sends a ReInvite message to user C, which carries SDP data to instruct user C to establish a media stream between users B and C and carries P-NOTIFICATION header field data to instruct a call transfer.
8. After AG2 (for user C) receives the ReInvite message for a call transfer, AG2 converts the message into a Notify message to notify user C.
9. The IMS sends a Notify message to AG1 to terminate the call transfer. Then user A exits the session, and user B talks with user C.

Implicit Transfer for ISDN Users (When User C Is Alerted)

If user C is alerted and then user A initiates a call transfer, the implicit transfer service is implemented as shown in Figure 1-70.

Figure 1-70 Service flow of implicit transfer for ISDN users (when user C is alerted)


UserA
SIP AG1
IMS
SIP AG2
UserB
UserC
UserA holds UserB, UserC answered the call request of UserA
1
Facility
2
Refer
202
Notify
200
3
Update
200
Update
200
Notify
Notify
4
Bye
200
Disconect
Release
ReleaseComplete
5
Bye
200
Disconect
Release
ReleaseComplete
6
200
Connect
Terminate
200 response
Ack
ConnectAck
7
ReInvite
8
200
Ack
Notify
9
ReInvite
10
200
Ack
Notify
11
Notify
200
UserA is not in any call, UserB and UserC are in call

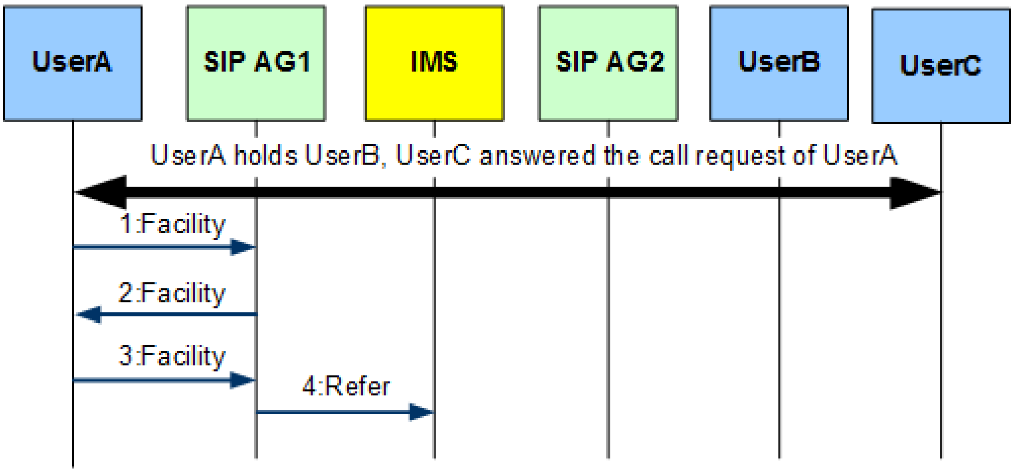
Compared with the flow described in Implicit Transfer for ISDN Users (When User C Answers a Call), this implicit transfer flow has 2 more steps as highlighted in Figure 1-70. The details are as follows:

1. During a call between users A and B, when C answers the call, user A sends a Facility message to AG1, instructing AG1 to transfer the call.
2. AG1 sends a Refer message to the IMS for a call transfer.
3. Before user C answers the call, the IMS sends an Update message to notify user B of a call transfer for user C.
4. The IMS releases the session between users A and B.
5. The IMS releases the session between users A and C.
6. The IMS terminates the 200 response that is sent when user C answers the call.
7. The IMS sends a ReInvite message to user B, which carries SDP data to instruct user B to establish a media stream between users B and C and carries P-NOTIFICATION header field data to instruct a call transfer.
8. After AG2 (for user B) receives the ReInvite message for a call transfer, AG2 converts the message into a Notify message to notify user B.
9. The IMS sends a ReInvite message to user C, which carries SDP data to instruct user C to establish a media stream between users B and C and carries P-NOTIFICATION header field data to instruct a call transfer.
10. After AG2 (for user C) receives the ReInvite message for a call transfer, AG2 converts the message into a Notify message to notify user C.
11. The IMS sends a Notify message to AG1 to terminate the call transfer. Then user A exits the session, and user B talks with user C.

Explicit Transfer for ISDN Users

Figure 1-71 shows how explicit transfer is implemented for ISDN users.

Figure 1-71 Service flow of explicit transfer for ISDN users



The service flow of explicit transfer for ISDN users is as follows:

1. During an ISDN call with user C, user A sends a Facility message to AG1, applying for ECTLinkId.
2. AG1 allocates an ECTLinkId and sends a Facility message to notify user A.
3. During an ISDN call with user B, user A sends a Facility message to AG1, instructing AG1 to transfer the call.
4. AG1 sends a Refer message to the IMS to instruct the IMS to transfer the call. The subsequent service flow is the same as that of implicit transfer.

1.6.7 Call Forwarding

The call forwarding (CF) service is a callee-side service. If a user has subscribed to the CF service and the call flow meets the call forwarding conditions, the call is forwarded to the preset forwarded-to party.

1.6.7.1 Call Forwarding

This topic describes the definition and benefits of the call forwarding (CF) service.

Definition

The CF service is a service on the terminating side. If a user has subscribed to the CF service and the call flow meets the call forwarding conditions, the call is forwarded to the preset forwarded-to party. Currently, the system supports the following call forwarding services:

- CFU: a terminating-side service that enables a service user to unconditionally forward all incoming calls to a preset forwarded-to number.
- CFB: a terminating-side service that enables a user to forward all incoming calls to a preset forwarded-to number when the user is busy on another call.
- CFNR: a terminating-side service that a user to forward all incoming calls to a designated forwarded-to number if the calls are not answered within a preset period.

Benefit

Beneficiary	Benefits
Carrier	The CF service enriches users' service experience, improves the network connection rate, and provides value-added services for carriers.
User	The CF service enables users to flexibly set the CF type to facilitate the selection of the terminal and the occasion for answering the calls and to avoid missing calls.

Standards Compliance

- ETSI 300 102-1
- EN300 207
- ETSI TS 124 504
- ETSI TS 183036
- ETSI TS 183043

- ITU-T Q.931

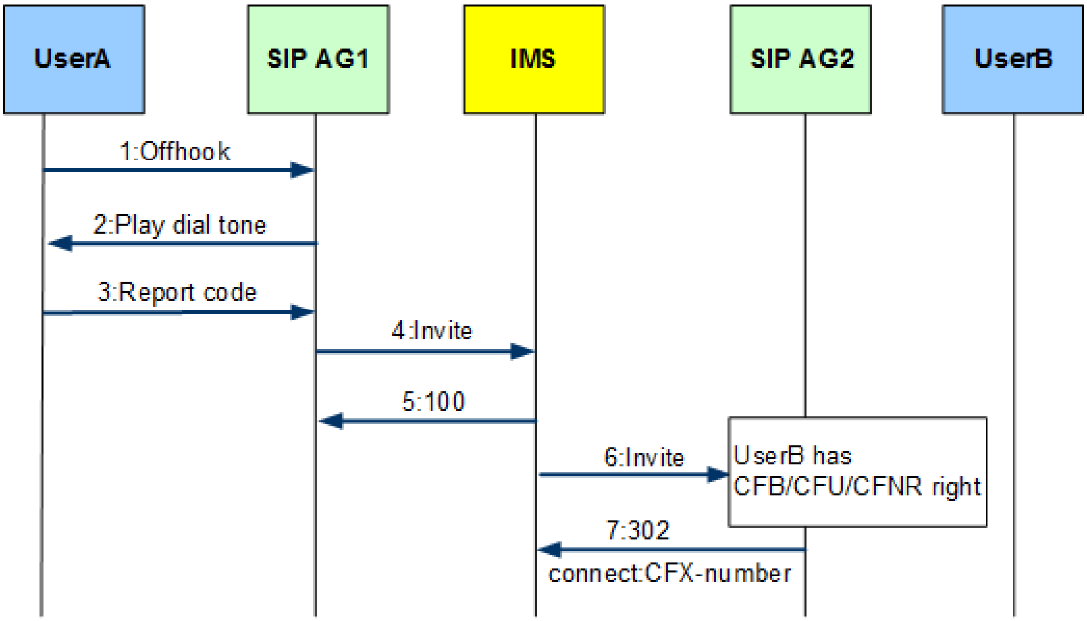
1.6.7.2 Call Forwarding Service Flow

This topic describes the call forwarding (CF) service flow for POTS users and ISDN users.

CF Service Flow for POTS Users

Figure 1-72 shows the call forwarding service flow for POTS users.

Figure 1-72 CF service flow for POTS users



The CF service flow is as follows:

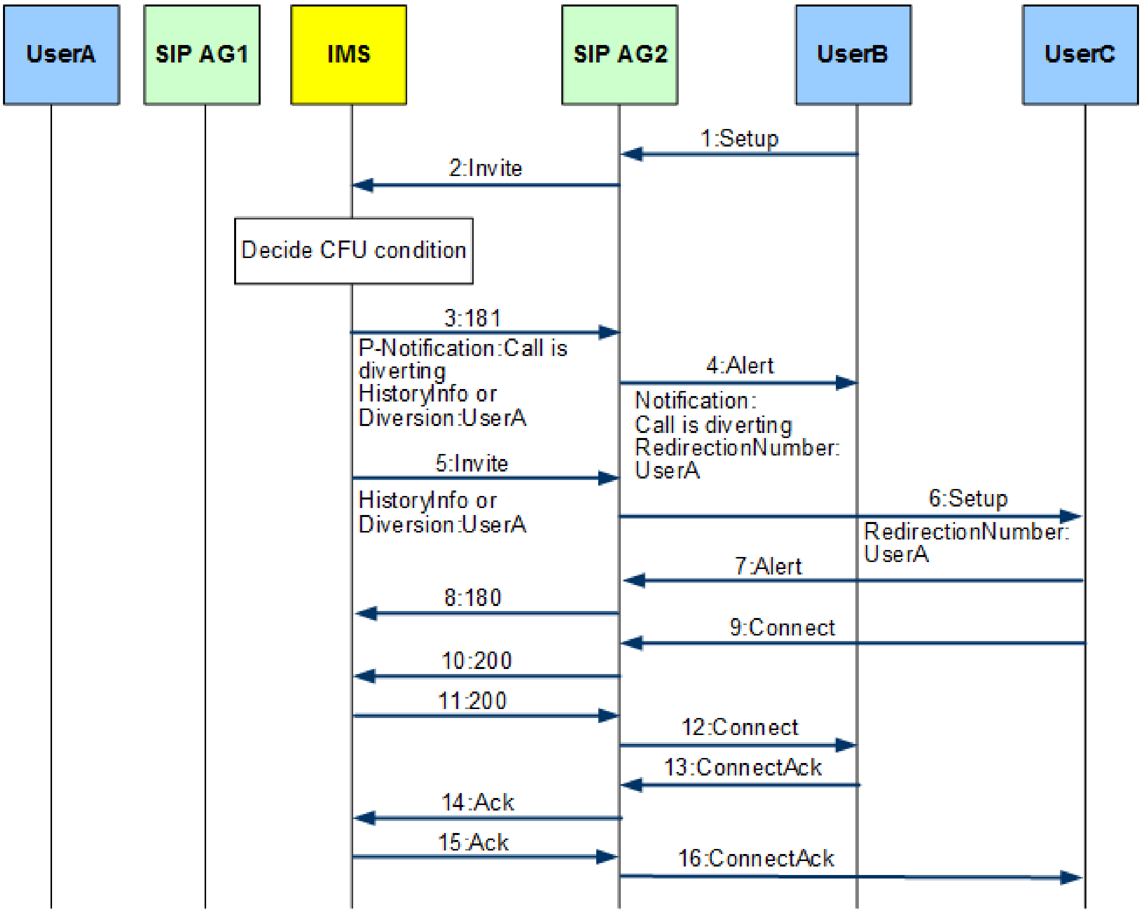
1. User A picks up the phone.
2. User A hears a dial tone.
3. User A reports the number information.
4. AG1 sends an Invite message to the IMS.
5. The IMS sends the 100 trying temporary response message.
6. The IMS sends the Invite message from user A.
7. AG2 determines that the current user meets the call forwarding conditions and returns a 302 message in which the Connect header field carries the forwarded-to number.

Then, the IMS initiates a call forwarding based on the forwarded-to number in AG2 response.

CFU Service Flow for ISDN users

Figure 1-73 shows the call forwarding unconditional (CFU) service flow for ISDN users.

Figure 1-73 CFU service flow for ISDN users



User A is a call forwarding service user. When user A receives a call from user B, the call is forwarded unconditionally to C without passing through user A. The CFU service flow is as follows:

1. The non-service user B initiates a call to service user A and sends a Setup message.
2. AG2 converts the Invite message and sends it to the IMS.
3. After receiving the message, the IMS determines and triggers the CFU service logic, and returns a 181 message carrying the **P-Notification:Call is diverting** notification to user B. The **History-Info** or **Diversion** header field is optional, and the call forwarding information is recorded.
4. AG2 converts the 181 message into an ISDN Alert message, carries the notification information unit, and generates the **RedirectionNumber** information unit on the ISDN side based on the **History-Info** or **Diversion** header field.
5. The IMS forwards the Invite message to user C based on the configuration. The Invite message carries the **History-Info** or **Diversion** header field and records the forwarding information.
6. AG2 converts the received Invite message into an ISDN Setup message and generates an ISDN **RedirectionNumber** information unit based on the **History-Info** or **Diversion** header field.
7. The forwarded-to party answers the call and sends a Connect message.

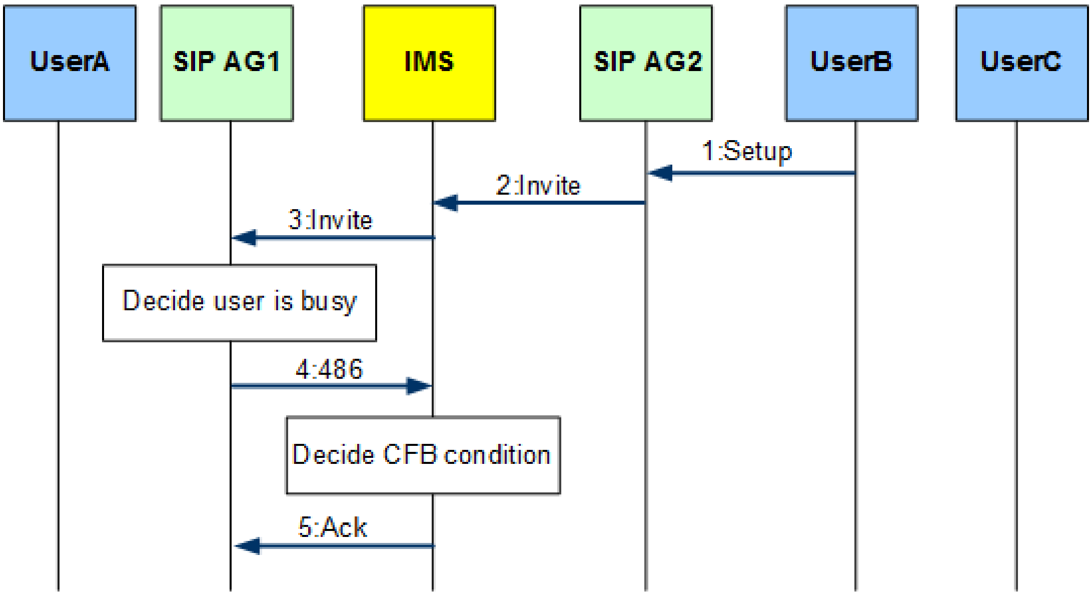
NOTE

The subsequent service flow is the same as the basic call flow. After the call is answered, non-service users B and C enter the conversation state.

CFB Service Flow for ISDN Users

The busy status in the call forwarding busy (CFB) service can be determined either by the IMS or the AG. When determined by the IMS, the CFB service flow is the same as the CFU service flow, except that the conditions for triggering the call forwarding flow are different. Figure 1-74 shows the call forwarding flow when the AG determines that the call is busy.

Figure 1-74 CFB service flow for ISDN users



User A is the CFB service party. The CFB service flow for ISDN users is as follows:

1. The non-service user B initiates a call to user A and sends a Setup message.
2. AG2 converts the Invite message and sends it to the IMS.
3. The IMS does not determine whether user A is busy, and sends the Invite message to AG1.
4. AG1 determines that user A is busy and sends a 486 response to the Invite request.
5. After receiving the 486 response, the IMS determines that the CFB condition is met and starts the call forwarding flow.

The subsequent flow is the same as that described in CFU Service Flow for ISDN users.

1.6.8 Emergency Call

A user can make an emergency call if the connected port is in remote block state and the SIP proxy server is normal and the dialed number matches the emergency digitmap.

1.6.8.1 Emergency Call

This topic describes the definition and benefits of the emergency call service.

Definition

When the user port is in the remote blocking state and the SIP proxy server is normal, if the number matches the emergency call digitmap, the emergency call can be initiated by dialing the emergency call numbers such as 110, 120, and 119.

Benefit

Beneficiary	Benefits
Carrier	The emergency call service brings added value and provides users with help in an emergency, improving user satisfaction.
User	Users can quickly dial preset emergency numbers in an emergency to help customers cope with the situation.

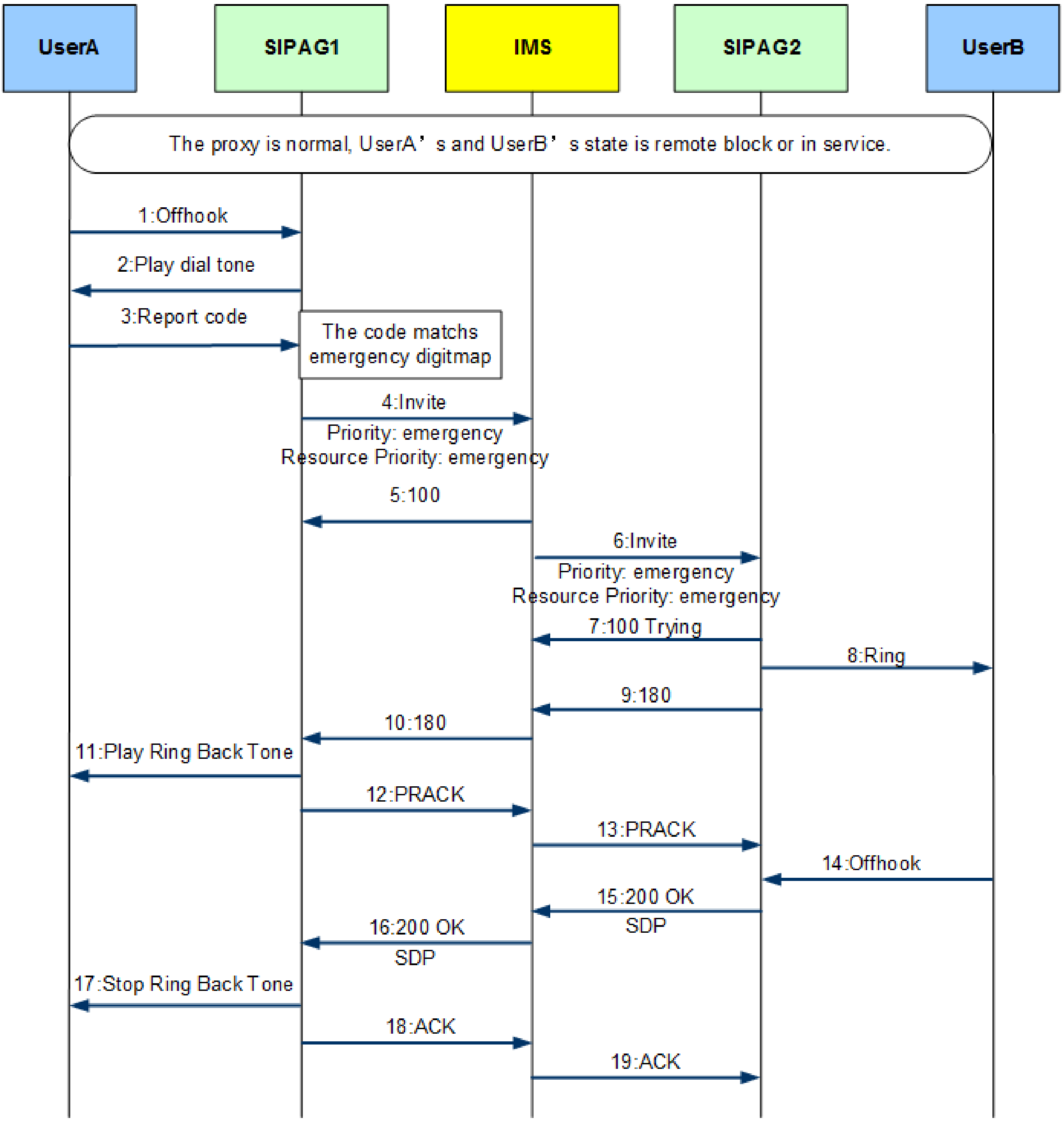
1.6.8.2 Emergency Call Service Flow

This topic takes user A as an example to describe the emergency call service flow.

POTS Emergency Call Service Flow

Figure 1-75 shows the emergency call service flow for POTS users.

Figure 1-75 Emergency call service flow for POTS users



This service flow is similar to a basic call flow. The differences are as follows:

- The service port may be in the remote blocking state (the proxy is normal). If the number matches the emergency digitmap, the call can be initiated.
- The initial Invite message carries the Priority and Resource Priority header fields, and the content is emergency.
- If the called party is in the remote blocking state (the proxy is normal), the system checks the content of the **Priority** and **Resource Priority** header fields. If the content is emergency, the call is allowed.

NOTE

The emergency call service flow for ISDN users is similar to the emergency call service flow of a POTS user.

1.6.9 Hotline

This service functions in this way: If the user who registers this service does not dial a number in a specified period (such as 5s) after picking up the phone, the system automatically connects this user to a fixed number (hotline number).

1.6.9.1 Hotline

This topic describes the definition and benefits of the hotline service.

Definition

This service functions in this way: If the user who registers this service does not dial a number in a specified period (such as 5s) after picking up the phone, the system automatically connects this user to a fixed number (hotline number). The hotline service can be classified into ordinary hotline and immediate hotline according to the no-dial interval.

- Ordinary hotline: Also called delay hotline. If the user does not dial a number in a specified period after picking up the phone, the system automatically connects this user to the hotline number.

NOTE

The delay hotline duration can be configured when configuring the service data.

- Immediate hotline: After the user picks up the phone, the system automatically connects this user to the hotline number.

Benefit

Beneficiary	Benefits
Carrier	The hotline service enriches user experience and enhances carriers' competitiveness.
User	When a user cannot dial the called number, the system automatically connects this user to a preset number to prevent unexpected events. This also saves the user from the trouble of dialing the called number again.

Standards Compliance

ETSI TS 183 010

3GPP TS 24610

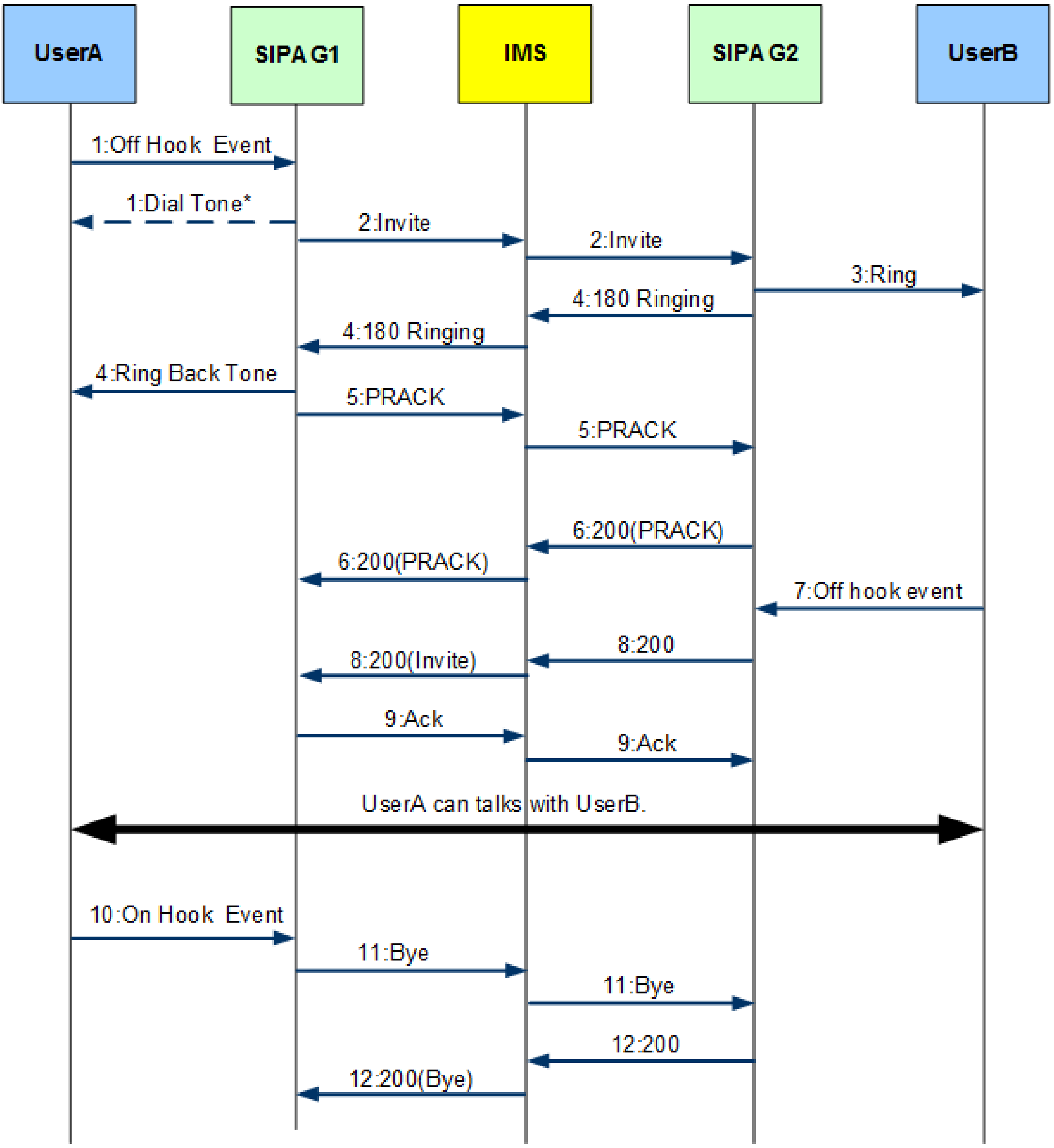
1.6.9.2 Hotline Service Flow

This topic takes service user A and common user B as an example to describe the hotline service flow.

POTS User Hotline Service Flow

Figure 1-76 shows the service flow of the POTS user hotline service.

Figure 1-76 POTS user hotline service flow



The hotline service flow is as follows:

1. User A (service party) picks up the phone and calls user B.

NOTE

* For the immediate hotline service, after a user picks up the phone, the dial tone is not played and a call is initiated immediately. For the delay hotline service, after a user picks up the phone, the dial tone is played. A hotline call is initiated if the user does not dial a number after the delay timer expires.

2. AG1 initiates a call to user B. The Invite message carries user B's number. AG2 receives the user A's call initiated by the IMS.
3. AG2 sends a ring tone to user B.

4. After user B's phone rings, AG2 sends a 180 response to user A. After receiving the 180 response message from the IMS, AG1 plays the ringback tone to user A.
5. AG1 sends a PRACK message to AG2.
6. AG1 receives the PRACK responses from AG2 and the IMS.
7. User B picks up the phone.
8. After detecting that user B picks up the phone, AG2 sends a 200 message to AG1.
9. After receiving the 200 message from the IMS, AG1 returns an Ack message to AG2. Then, a call is set up between users A and B.
10. User A hangs up after the call ends.
11. After detecting that user A hangs up, AG1 sends a Bye message, asking the IMS to release the call. After receiving the BYE message, the IMS sends a Bye message to user B (called party) and releases the call.
12. After receiving the Bye message, AG2 releases the call and returns a 200 OK response to the IMS. AG1 receives a 200 OK response to the Bye message. The call is released.

NOTE

The service flow of the ISDN user hotline service is similar to that of the POTS user hotline service.

1.6.10 Message Waiting Indication

This is a message prompt service, with which, the voice mail system (VMS) or unified message system (UMS) notifies users of changed number or status of messages, including emails, short messages, faxes, and leaving messages.

1.6.10.1 Message Waiting Indication

This topic describes the definition and benefits of the message waiting indication (MWI) service.

Definition

The MWI service is a message prompt service. When the status of a message (including email, SMS, fax, and voice message) on the voice mail server (VMS) changes, the MWI service sends a notification to the user, prompting the user to view the messages in a timely manner.

The MWI service permission can be configured locally or obtained through subscription. After the user has the MWI service permission, the user can turn on or off the indicator according to the instruction of the IMS. Whether the MWI has a ringing alert is controlled by the system parameter. The message MWI signal delivery can be delayed in the off-hook state.

Benefit

Beneficiary	Benefits
Carrier	The MWI service enhances user experience and carriers' competitiveness.
User	With the MWI service, the user can know the change of the information stored in the VMS in time.

Standards Compliance

RFC3323

EN300 745

ETSI TS 183 006

ETSI TS 124 406

ETSI TS 300 745

ETSI TS 183 036

3GPP TS 24606

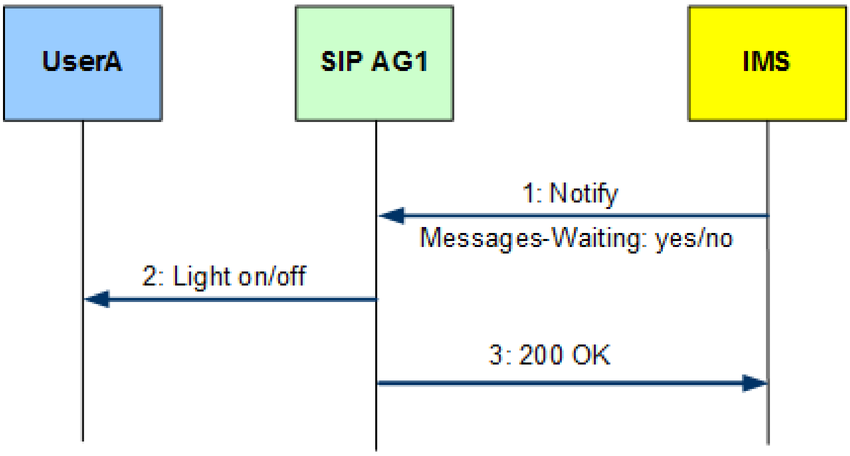
1.6.10.2 Service Flow of Message Waiting Indication

This topic describes the message waiting indication (MWI) service flow for POTS and ISDN users.

MWI Service Flow for POTS Users

Figure 1-77 shows the MWI service flow for POTS users.

Figure 1-77 MWI service flow for POTS users



User A is a service party and has subscribed to the MWI service. The MWI service flow for POTS users is as follows:

1. When detecting that the VMS event status changes, the IMS sends a Notify message to notify AG1. If the value of **Messages-Waiting** field is **yes**, the MWI status is on. If the value is **no**, the MWI status is off.
2. After receiving the Notify message, AG1 sends a signal to user A, indicating whether to turn on or off the WMI.
3. AG1 responds with a 200 OK message.

MWI Service Flow for ISDN Users

After the MWI status of an ISDN user changes, the AG notifies the terminal of the change in either of the following modes: immediate mode and delayed mode.

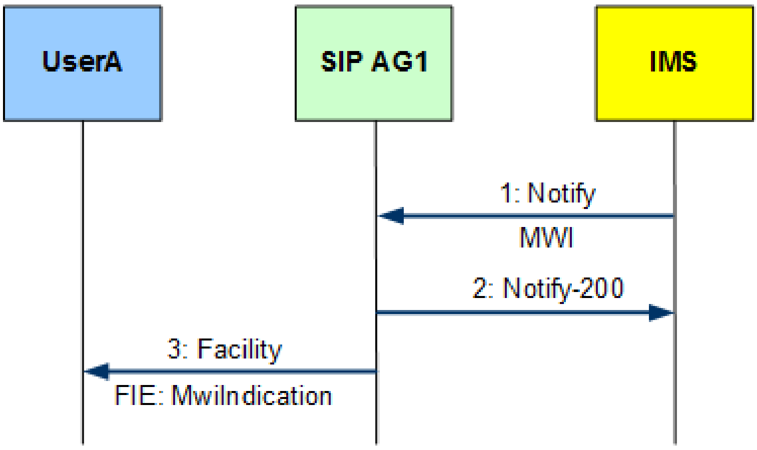
- In immediate mode, when the AG receives the MWI status change message from the IMS, the AG immediately sends a Facility message to notify the terminal.
- In delayed mode, after receiving the MWI status change message from the IMS, the AG buffers the message until the terminal sends a Setup message, then sends a Facility message to notify the terminal of the MWI status.

The following describes the details.

MWI (immediate mode) Service Flow for ISDN Users

Figure 1-78 shows the MWI (immediate mode) service flow for ISDN users.

Figure 1-78 MWI (immediate mode) service flow for ISDN users



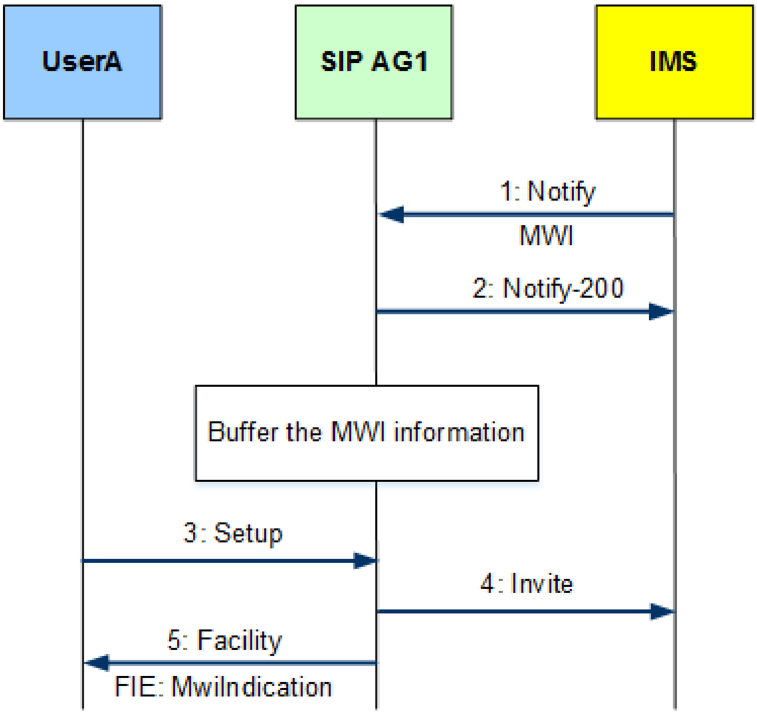
User A is a service party and has subscribed to the MWI service. The MWI (immediate mode) service flow for ISDN users is as follows:

1. When detecting that the VMS event status changes, the IMS sends a Notify message to notify AG1 of the event.
2. After receiving the Notify message, AG1 returns a Notify 200 message.
3. AG1 sends a Facility message, in which the FIE message carries the MWI information of the user.

MWI (delayed mode) Service Flow for ISDN Users

Figure 1-79 shows the MWI (delayed mode) service flow for ISDN users.

Figure 1-79 MWI (delayed mode) service flow for ISDN users



User A is a service party and has subscribed to the MWI service. The MWI (delayed mode) service flow for ISDN users is as follows:

1. When detecting that the VMS event status changes, the IMS sends a Notify message to notify AG1 of the event.
2. After receiving the Notify message, AG1 returns a Notify 200 message. AG1 determines that user A has subscribed to the MWI (delayed mode) service and saves the MWI information in the buffer.
3. User A sends a message reading request.
4. After receiving the request from user A, AG1 sends an Invite message to the IMS.
5. AG1 sends a Facility message, in which the FIE message carries the MWI information of the user.

NOTE

When the service party is a POTS user, the AG delivers the MWI signal to the user. When the service user is an ISDN user, the AG sends a Q931 message to the user. The Facility IE in the Q931 message carries the MWI signal.

1.6.11 Calling Line Identification Presentation

The calling line identification presentation (CLIP) service enables the AG to determine whether to display the identity and name of the calling party to the called party based on the subscription data of a user during session setup.

1.6.11.1 Calling Line Identification Presentation

This topic describes the definition and benefits of the calling line identification presentation (CLIP) service.

Definition

The calling line number presentation service can be classified into the following types:

- The CLIP service allows the calling number, name, and other information to be presented to the called party when the called party is alerted.
- The calling line identification restriction (CLIR) service enables the calling party to restrict the presentation of the number, name, and multimedia information to the called party during call setup if the calling party has subscribed to the CLIR service and the called party has not subscribed to the RIO service.
- The calling line identification restriction override (RIO) service is a callee-side service that allows the number of the calling party to be presented to the called party even if the calling party registers CLIR.

Benefit

Beneficiary	Benefits
Carrier	The CLIP service enriches users' service experience, improves the network connection rate, and provides value-added services for carriers.
User	The CLIP service enables users to flexibly set the display type of the calling number to facilitate the selection of the terminal and the occasion for answering the calls and to avoid missing calls.

Implementation

The CLIP service does not require service logic and does not need to be configured on the device. After subscribing to the basic call service, voice users can subscribe to and use the CLIP service on the terminal.

Standards Compliance

ETSI EN 300 092-1

ETSI EN 300 093-1

RFC3323

ETSI TS 183 036

ETSI TS 183 043

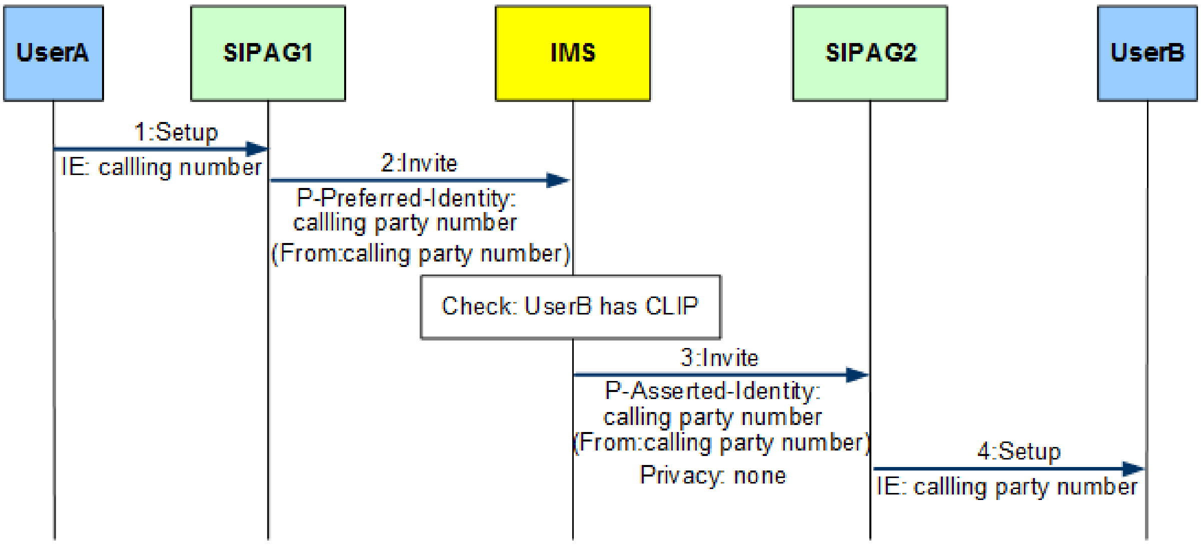
1.6.11.2 Service Flow of Calling Line Identification Presentation

This topic describes the calling line identification presentation (CLIP) service flow for POTS and ISDN users, including CLIP, calling line identification restriction (CLIR), and calling line identification restriction override (RIO) service flows.

CLIP Service Flow for ISDN Users

Figure 1-80 shows the CLIP service flow for ISDN users.

Figure 1-80 CLIP service flow for ISDN users



The CLIP service flow for ISDN users is as follows (taking user A calling user B as an example):

1. User A initiates a call. The Setup message contains the **calling party number** information element.
2. AG1 identifies the **calling party number** information element in the Setup message and maps it to the **P-Preferred-Identity** or **From** header field.
3. After receiving the Invite message, the IMS determines the service permission of user A and B, maps the service permission of users A and B to a corresponding message, and sends the message to AG2.
   - If user A has the CLIP service permission or user B has the RIO service permission, the IMS removes the **P-Preferred-Identity** header field from the message, adds the **P-Asserted-Identity** and **Privacy** header fields to the message, and keeps the **From** header unchanged. If the value of the **Privacy** header field is **none**, it indicates that the calling number information is displayed to the called party.
   - If user A has the CLIR service permission and B does not have the RIO service permission, the IMS removes the **P-Preferred-Identity** header field from the message, adds the **P-Asserted-Identity** and **Privacy** header fields to the message, keeps the **From** header field unchanged, and sets the value of the **Privacy** header field to **id**, **header**, or **user**, indicating that the calling number information is not displayed to the called party.
4. AG2 determines whether the Setup message sent to user B carries the calling number information element based on the **P-Asserted-Identity**, **From**, and **Privacy** header fields in the message.
   - If the **Privacy** header field is set to **none** or does not exist, the Setup message sent from AG2 to user B carries the **calling party number** information element.
   - If the value of the **Privacy** header field to **id**, **header**, or **user**, the Setup message sent from AG2 to user B does not carry the **calling party number** information element.

NOTE

The CLIP service flow for a POTS user is similar to that of an ISDN user.

1.6.12 Connected Line Identification Presentation

The connected line identification presentation (COLP) service enables the AG to determine whether to display the called party's identifier to the calling party based on the subscription data of a user during session setup and to process the call.

1.6.12.1 Connected Line Identification Presentation

This top describes the definition and benefits of the Connected Line Identification Presentation (COLP) service.

Definition

The COLP service determines whether to display the called party's identifier to the calling party based on the subscription data of the user and performs corresponding operations. The called line number presentation service can be classified into the following types:

- COLP: a service on the originating side. When the called party is not subscribed to the COLR service, the called number is displayed on the calling party's terminal when the called party answers the call.
- Connected Line Identification Restriction (COLR): a service on the terminating side. If the called party is subscribed to the COLR service, the called number is not displayed to the calling party even if the calling party has subscribed to the COLP service.

Benefit

Beneficiary	Benefits
Carrier	The COLP service enriches users' service experience, enhances carriers' competitiveness, and provides value-added services for carriers.
User	The COLP service enables users to set the CLIP service by themselves and experience different services. • The called party can determine whether to display the called number to the calling party in the session connection phase. • The calling party can determine whether to display the called number in the session connection phase.

Standards Compliance

EN300 356

EN300 097

RFC4916

EN300 098

ETSI TS 183 036

ETSI TS 183 043

ETSI TS 300 097

ETSI TS 300 098

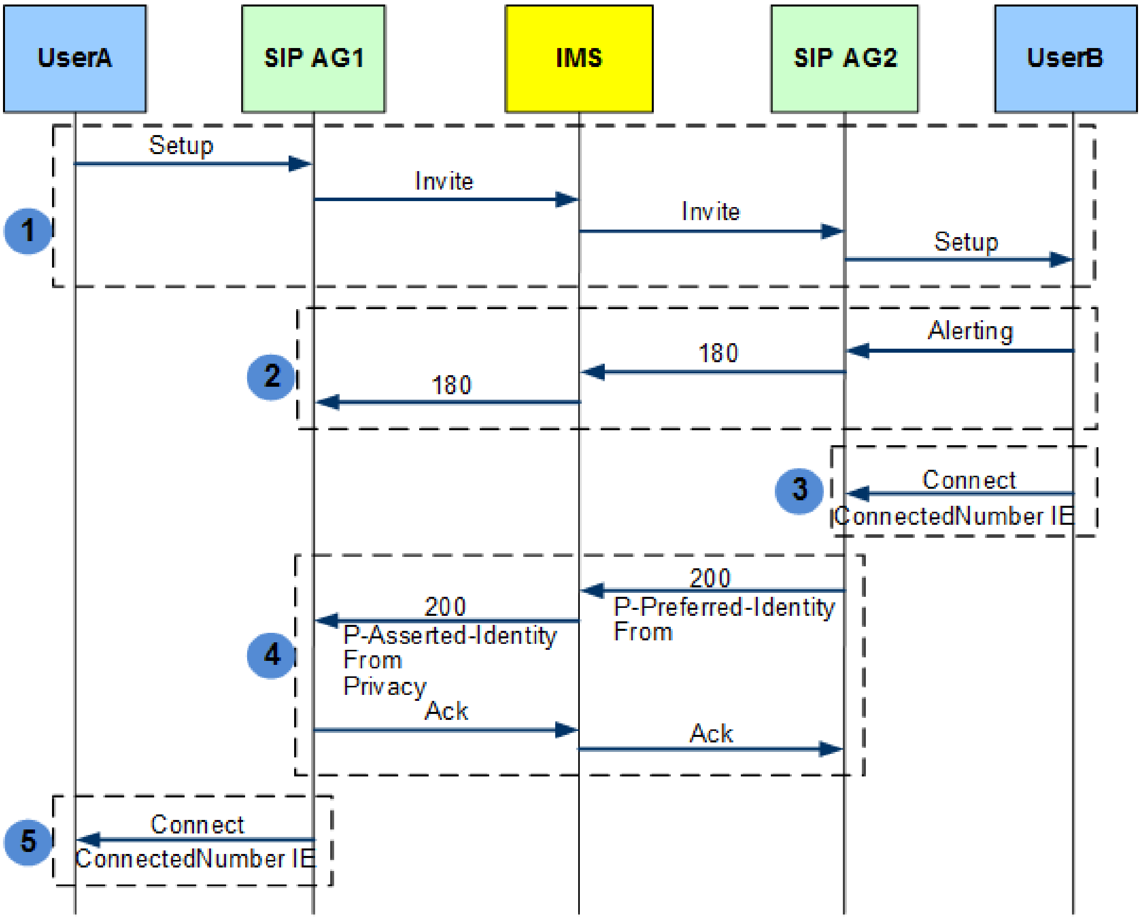
1.6.12.2 Service Flow of Connected Line Identification Presentation

This topic describes the connected line identification presentation (COLP) service flows for POTS and ISDN users.

COLP Service Flow for ISDN Users

Figure 1-81 shows the COLP service flow for ISDN users.

Figure 1-81 COLP service flow for ISDN users



User A is the calling party, and user B is the called party. The COLP service flow for ISDN users is as follows:

1. User A initiates a call to user B.
2. User B sends a 180 message to user A.
3. User B sends a Connect message, which carries the ConnectedNumber information element, indicating the called number.
4. AG2 maps the Connect message to a 200 response and sends it to the IMS. The 200 response carries the **P-Prefered-Identify** or **From** header field. After receiving the 200 message, the IMS maps the **P-Prefered-Identify** header field in the 200 message to the **P-Asserted-Identify** header field based on the COLP service permission of users A and

B, and adds the **Privacy** header field to instruct AG1 whether to display the called number.

5. The user A receives a Connect message from AG1 and displays the information based on the **Privacy** header field in the Connect message.

NOTE

The COLP service flow of a POTS user is similar to that of an ISDN user.

1.6.13 Callback

When user A calls user B and user B is busy or does not answer the call, the AG automatically sets up a call if the user has subscribed to the callback service and meets the callback conditions. The callback service includes the Completion of Calls to Busy Subscriber (CCBS) and Completion of Communication on No Reply (CCNR) services.

1.6.13.1 Callback

This topic describes the definition and benefits of the callback service.

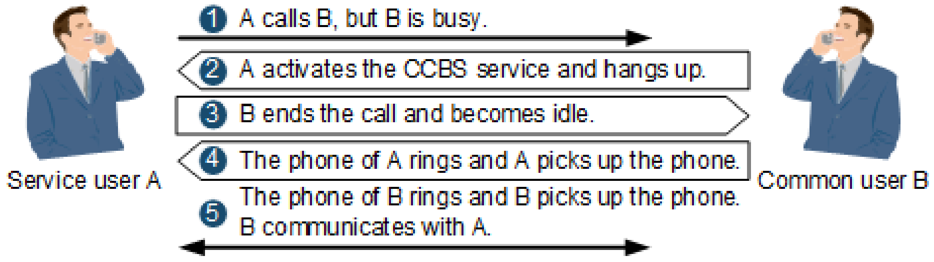
Definition

The callback service includes the Completion of Calls to Busy Subscriber (CCBS) and Completion of Communication on No Reply (CCNR) services.

The following describes the CCBS service: When user A (calling party) calls user B (called party) and user B is busy, user A can activate the CCBS service if user A has registered the service. When user B is idle, the AG automatically calls users A and B back to set up a call.

Assume that user A has subscribed to the CCBS service, and user B is a common user. Figure 1-82 shows the schematic diagram.

Figure 1-82 Schematic diagram of the CCBS service



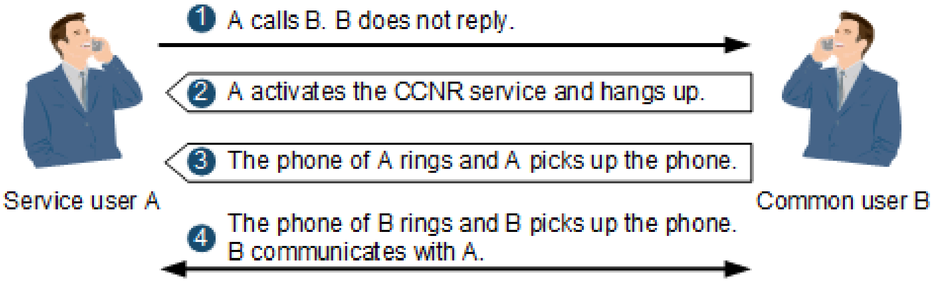
1 A calls B, but B is busy.
2 A activates the CCBS service and hangs up.
3 B ends the call and becomes idle.
4 The phone of A rings and A picks up the phone.
5 The phone of B rings and B picks up the phone. B communicates with A.
Service user A
Common user B

1. User A calls B. User B is busy.
2. User A activates the CCBS service and hangs up.
3. User B becomes idle after a call ends.
4. User A picks up the phone after the phone rings.
5. User B picks up the phone after the phone rings. User A talks with user B.

The following describes the CCNR service: When user A (calling party) calls user B (called party) and user B does not answer the call, user A can activate the CCNR service if user A has registered the service. When user B is idle, the AG automatically calls users A and B back to set up a call.

Assume that user A has subscribed to the CCNR service, and user B is a common user. Figure 1-83 shows the schematic diagram.

Figure 1-83 Schematic diagram of the CCNR service


1 A calls B. B does not reply.
2 A activates the CCNR service and hangs up.
3 The phone of A rings and A picks up the phone.
Service user A
Common user B
4 The phone of B rings and B picks up the phone. B communicates with A.

1. User A picks up the phone and calls user B. User B does not answer the call.
2. User A activates the CCNR service and hangs up.
3. User A picks up the phone after the phone rings.
4. User B picks up the phone after the phone rings. User A talks with user B.

Benefit

Beneficiary	Benefits
Carrier	The callback service improves the connection rate, enriches user service experience, enhances the carrier's competitiveness, and adds value to carrier services.
User	After the calling party activates the CCBS or CCNR service, the AG automatically detects the network status of the called party, which facilitates the calling party and improves user experience.

Implementation

The call back service does not require service logic or device configuration. Voice users can use the basic call service only after subscribing to it.

Standards Compliance

- ETSI TS 300 359
- ETSI TS 124 642
- ETSI TS 183 036

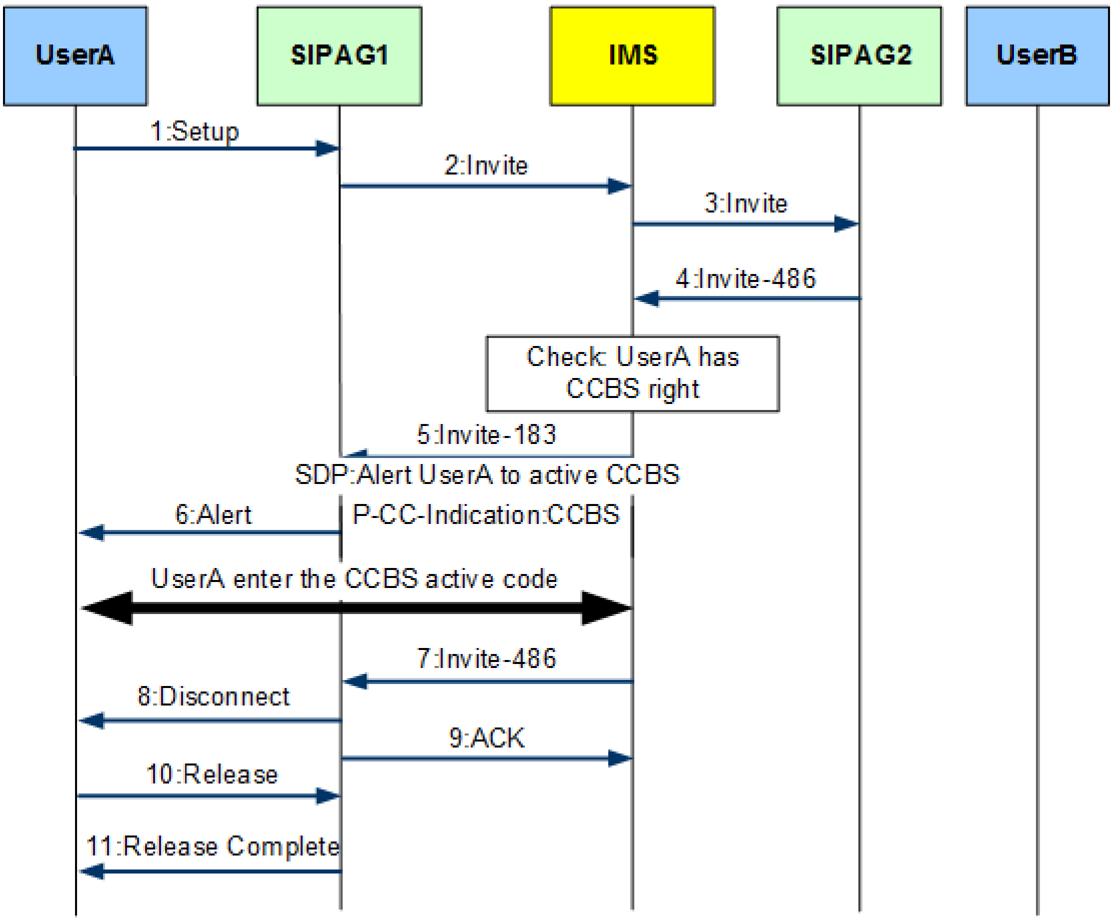
1.6.13.2 Callback Service Flow

This topic takes user service A and common user B as an example to describe the callback service flow, including the Completion of Calls to Busy Subscriber (CCBS) and Completion of Communication on No Reply (CCNR) service flows.

ISDN CCBS Service Flow

Figure 1-84 shows the service flow of the ISDN CCBS service.

Figure 1-84 ISDN CCBS service flow



User A is a callback service user. User A calls user B, and user B is busy. The service flow of activating the CCNR service by an ISDN user is as follows:

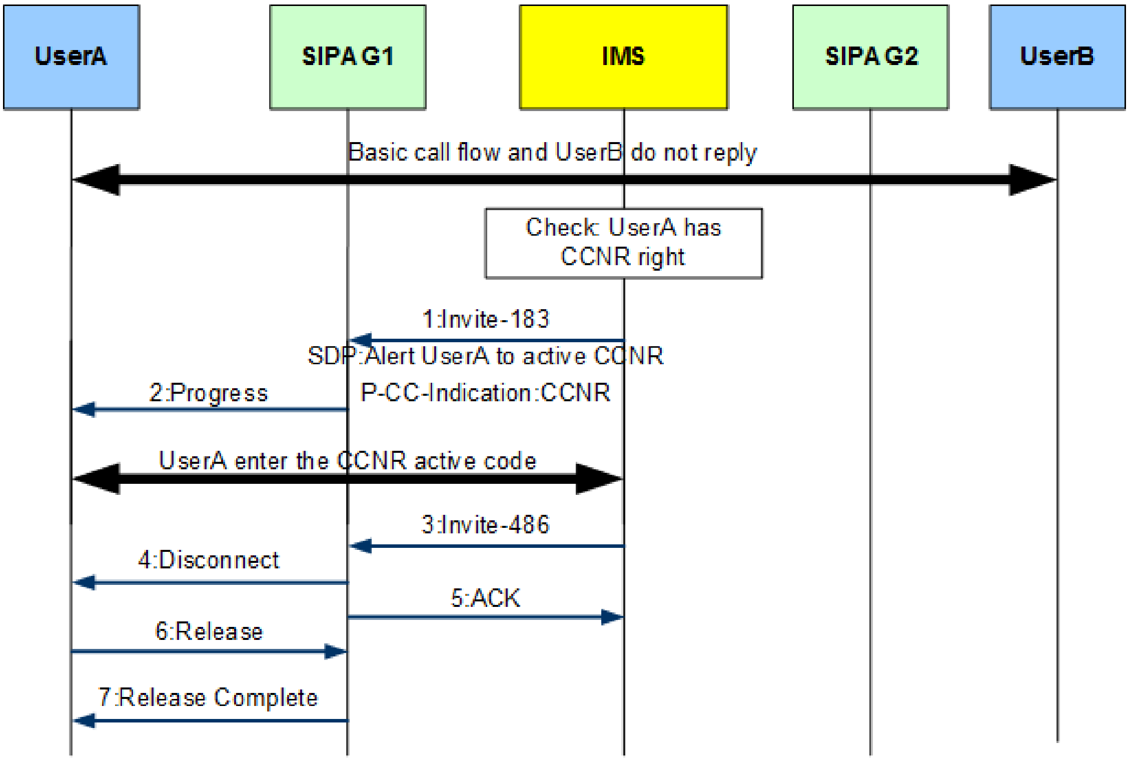
1. User A picks up the phone and calls user B.
2. AG1 sends an Invite message to user B.
3. The IMS sends an Invite message to user B.
4. AG2 replies with an Invite-486 message when user B is busy.
5. After detecting that user A has the CCBS service permission, the IMS sends an Invite-183 message to ask user A to activate the CCBS service.
6. After receiving the Invite-183 message, AG1 asks user A to activate the CCBS service. After receiving an Alert message, A enters the CCBS activation code.
7. After user A activates the CCBS service, the IMS sends an Invite-486 message to AG1, indicating that the call cannot be connected.
8. AG1 sends a Disconnect message to user A, asking user A to release the call.
9. AG1 returns an Ack message in response to the Invite-486 message sent from the IMS.
10. A hangs up and releases the call.

11. AG1 confirms that the call is released.

ISDN CCNR Service Flow

Figure 1-85 shows the service flow of the ISDN CCNR service.

Figure 1-85 ISDN CCNR service flow


UserA
SIPAG1
IMS
SIPAG2
UserB
Basic call flow and UserB do not reply
Check UserA has CCNR right
1:Invite-183
SDP:Alert UserA to active CCNR
P-CC-Indication:CCNR
2:Progress
UserA enter the CCNR active code
3:Invite-486
4:Disconnect
5:ACK
6:Release
7:Release Complete

User A is a callback service user. User A calls user B, and user B does not answer the call for a long time. The service flow of activating the CCNR service by an ISDN user is as follows:

1. After detecting that user A has the CCNR service permission, the IMS sends an Invite-183 message to ask user A to activate the CCNR service.
2. AG1 asks user A to activate the CCNR service.
3. After user A activates the CCNR service, the IMS sends an Invite-486 message to AG1, indicating that the call cannot be connected.
4. AG1 sends a Disconnect message to user A, asking user A to release the call.
5. AG1 returns an Ack message in response to the Invite-486 message sent from the IMS.
6. A hangs up and releases the call.
7. AG1 confirms that the call is released.

NOTE

The service flow of the POTS user callback service is similar to that of the ISDN user emergency call service.

1.6.14 Malicious Call Identification

A user who registers this callee-side service can identify the calling number if the user receives a malicious call or unsolicited call, such as a prank call or a phishing call.

1.6.14.1 Malicious Call Identification

This topic describes the definition and benefits of the malicious call identification (MCID) service.

Definition

The MCID service enables a user to identify the calling number when the user receives a malicious call or unsolicited call.

Benefit

Beneficiary	Benefits
Carrier	The MCID service enhances carriers' competitiveness and provides new revenue streams.
User	In the following cases, the called party can obtain the calling number of the calling party through the MCID service: • The called party has not subscribed to the originating identification presentation (OIP) service. • The calling party has subscribed to the originating identification restriction (OIR) service. • After the phone rings, the calling number is not displayed but the calling party hangs up. In addition, when a user answers a malicious call (for example, blackmail or extortion), the user can notify the police through the MCID service.

Standards Compliance

EN300 130

ETSI TS 300 130

ETSI TS 124 516

ETSI TS 183 016

ETSI TS 183 036

ETSI TS 183 043

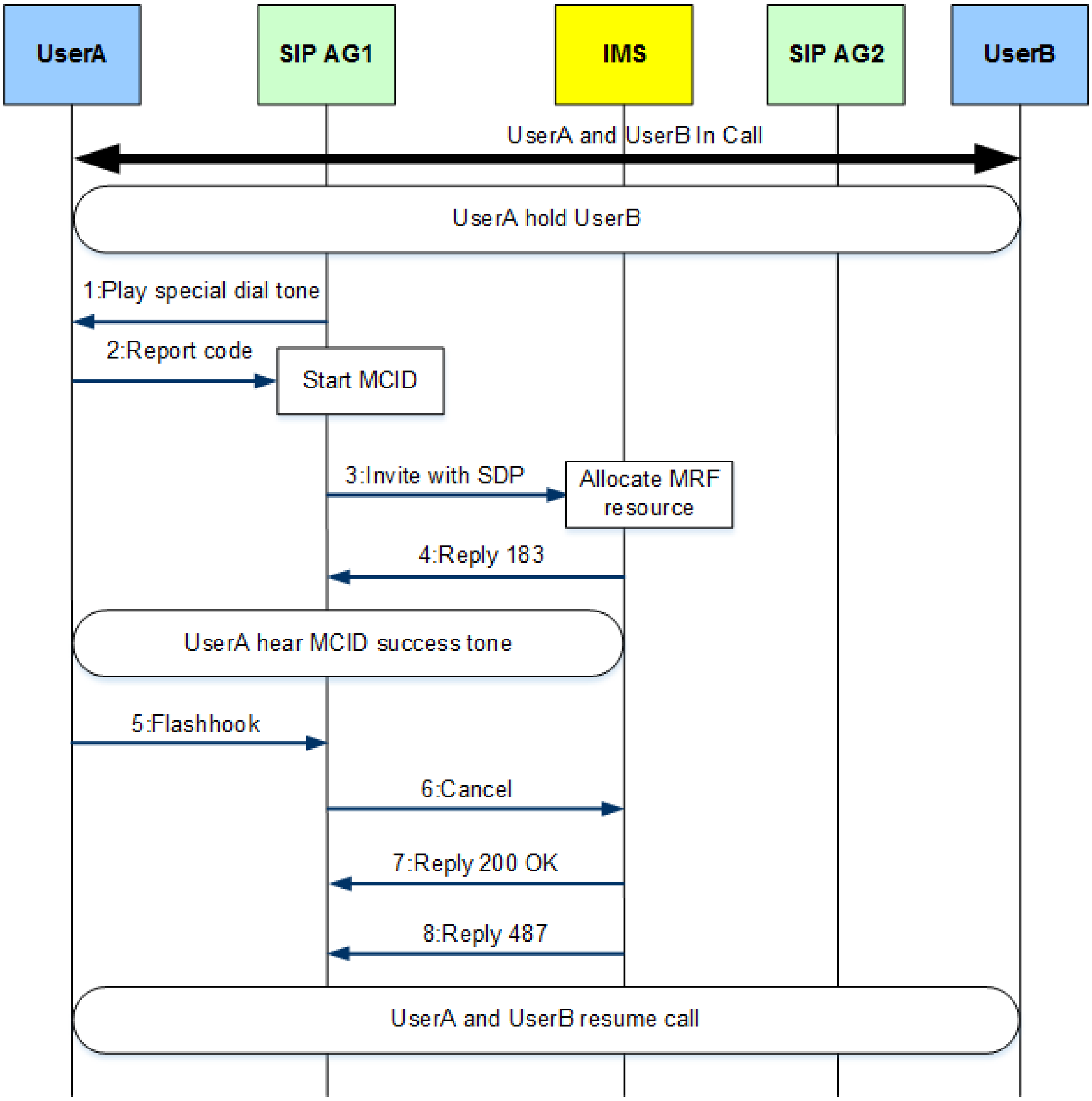
1.6.14.2 Service Flow of Malicious Call Identification

This topic takes the service user A and common user B as an example to describe the malicious call identification (MCID) service flows.

MCID Service Flow for POTS Users

The MCID service of a POTS user can be implemented in two ways: During a call, the user can dial the service access code to obtain the malicious call information by pressing the hookflash or dialing the service access code after the call ends (or after refusing the call), as shown in Figure 1-86 and Figure 1-87.

Figure 1-86 MCID service flow for POTS users during a call

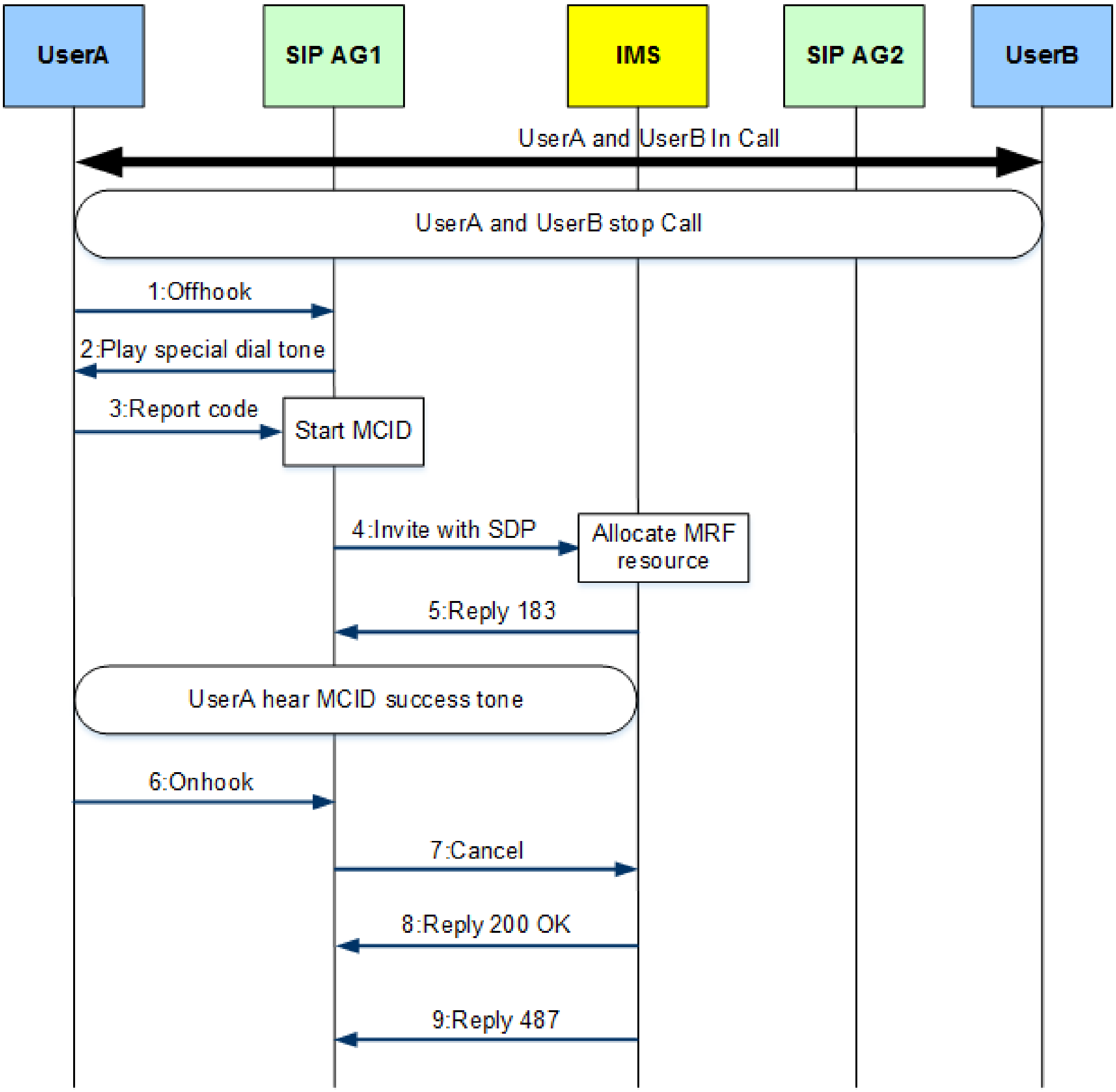


User A is a service user. User A presses hookflash, and user B is placed on hold. The MCID service flow for POTS users is as follows:

1. AG1 plays the special dial tone to user A.
2. User A calls the SCC number.
3. AG1 sends an unsolicited Invite message to the IMS.

4. The IMS allocates MRF resources and returns a 183 response. User A hears an announcement indicating that the MCID service is successful.
5. User A presses hookflash.
6. AG1 sends a Cancel message to cancel the previous unsolicited Invite request.
7. The IMS responds with a 200 OK message.
8. The IMS responds with an Ack message. The call between users A and B is resumed.

Figure 1-87 MCID service flow for POTS users after a call



User A is a service user. User A ends a call with user B. The MCID service flow for a POTS user is as follows:

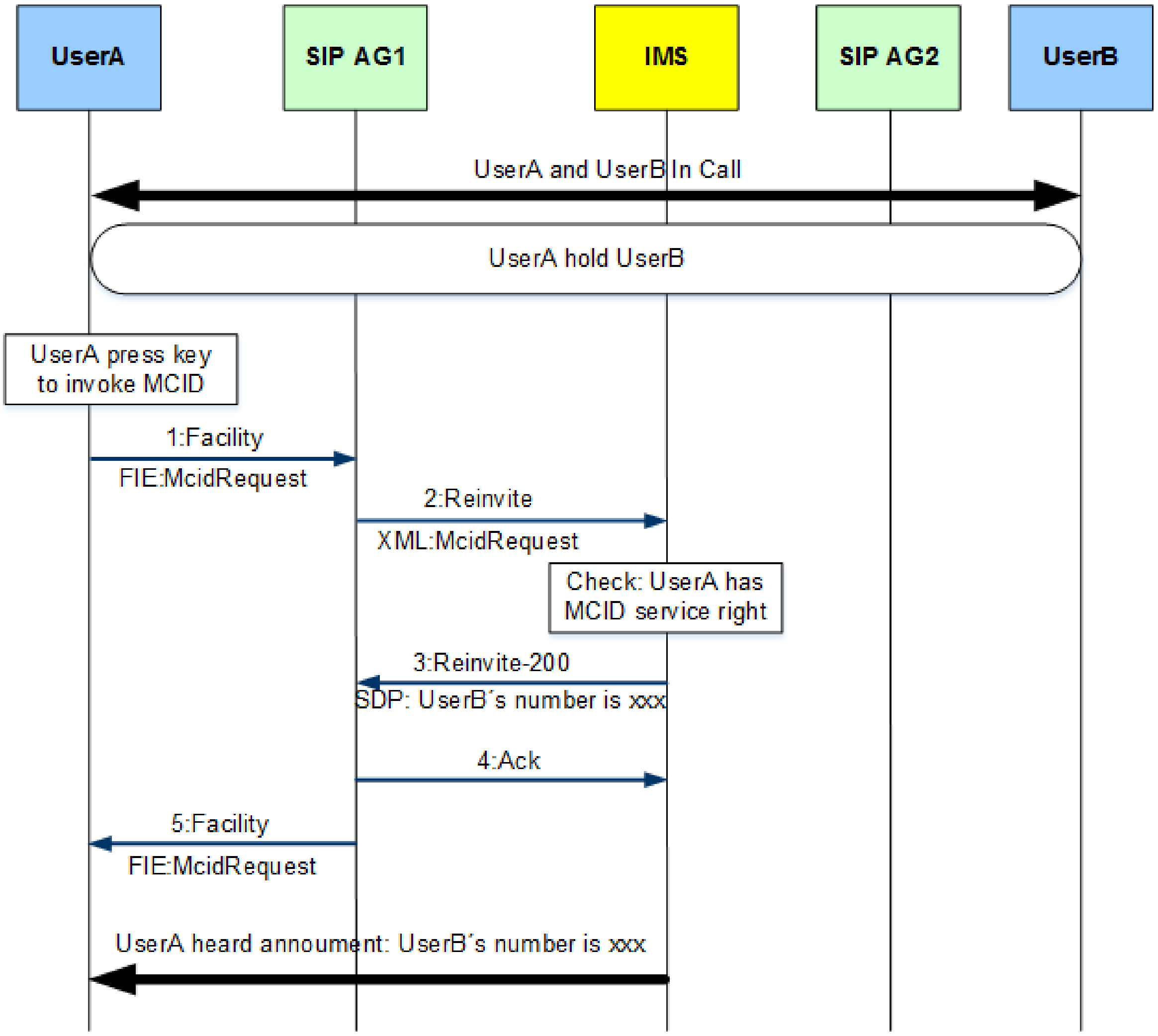
1. User A picks up the phone.
2. AG1 plays the special dial tone to user A.
3. User A calls the SCC number.
4. AG1 sends an unsolicited Invite message to the IMS.

5. The IMS allocates MRF resources and returns a 183 response. User A hears an announcement indicating that the MCID service is successful.
6. User A hangs up the phone.
7. AG1 sends a Cancel message to cancel the previous unsolicited Invite request.
8. The IMS responds with a 200 OK message.
9. The IMS responds with an Ack message. User A releases the call.

MCID Service Flow for ISDN Users

Figure 1-88 shows the service flow of the MCID service for ISDN users.

Figure 1-88 MCID service for ISDN users



Users A and B enter the conversation state through a call flow. User A is the service party (called party). The MCID service flow for ISDN users is as follows:

1. User A performs operations on the ISDN terminal and starts the MCID. The terminal sends a Facility message carrying the FIE message unit, indicating that the message is an MCID request.

2. AG1 maps the FIE to the XML message body and initiates media renegotiation to the IMS.
3. The IMS determines that A has the MCID service permission, and then notifies AG1 of the phone number of malicious caller through the SDP in media renegotiation.
4. After receiving the SDP message from the IMS, AG1 sends an Ack message.
5. AG1 sends a Facility message carrying the FIE message unit to user A, indicating the MCID result. User A hears user B's number through the terminal announcement.

1.6.15 Configuring the SIP Value-added Service

The SIP value-added service enriches user experience and brings value-added services for carriers. This section describes how to configure the SIP value-added service.

1.6.15.1 Configuring the Customized SIP Value-added Service Logic

The SIP value-added service is implemented through the service logic of the system. If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.

Prerequisites

- The customized logic file is available.
- The following configurations have been completed:
  a. Configuring the Upstream VLAN Interface
  b. Configuring the Media and Signaling IP Address Pools
  c. Adding an SIP Interface

Procedure

Step 1 In config mode, run the **load sip-srvlogic** command to load the customized service logic.

Step 2 In SIP mode, run the **if-sip attribute basic srvlogic-index 5** command to make the customized service logic file take effect.

----End

Example

To load the customized service logic file (file name sip_service_logic) on the server whose IP address is 10.13.0.10 through FTP, and make the file effective on the added SIP interface 0, run the following commands:

```
huawei(config)#load sip-srvlogic ftp 10.13.0.10 sip_service_logic
huawei(config)#interface sip 0
huawei(config-if-sip-0)#if-sip attribute basic srvlogic-index 5
```

1.6.15.2 Configuring the Call Waiting Service

The call waiting (CW) service enriches user experience and brings value-added services to carriers. This section describes how to configure the CW service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the call waiting permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the call waiting permission for a service user.

----End

Example

To configure the CW service for POTS user A with the listed data planning, run the following commands:

- The CW service code of the POTS user is 5, and the service priority is 1 (the default value is used as an example).
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 5 1
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 cw enable
```

To configure the CW service for ISDN user A with the listed data planning, run the following commands:

- The CW service code of the BRA user is 11, and the service priority is 11 (the default value is used as an example).

- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 11 11
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 cw enable
```

1.6.15.3 Configuring the Call Hold Service

The call hold (CH) service enriches user experience and brings value-added services to carriers. This section describes how to configure the CH service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the call hold permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the call hold permission for a service user.

----End

Example

To configure the CH service for POTS user A with the listed data planning, run the following commands:

- The CH service code of the POTS user is 26, and the service priority is 26 (the default value is used as an example).

- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 26 26
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 callhold enable
```

To configure the CH service for ISDN user A with the listed data planning, run the following commands:

- The CH service code of the BRA user is 9, and the service priority is 9 (the default value is used as an example).
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 9 9
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 callhold enable
```

1.6.15.4 Configuring the Local Three Party Conference Call Service

The local three party conference call (3PTY) service enriches user experience and brings value-added services to carriers. The following describes the procedure for configuring the local 3PTY service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In SIP mode, run the **mg-software parameter** command to set the 3PTY service mode.

Step 3 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 4 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the local 3PTY permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the local 3PTY permission for a service user.

----End

Example

To configure the local 3PTY service for POTS user A with the listed data planning, run the following commands:

- The local 3PTY service code of the POTS user is 1, and the service priority is 5 (the default value is used as an example).
- The 3PTY service mode: local 3PTY
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 1 5
huawei(config-if-sip-0)#mg-software parameter 28 0
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 three-party
enable
```

To configure the local 3PTY service for ISDN user A with the listed data planning, run the following commands:

- The local 3PTY service code of the BRA user is 7, and the service priority is 7 (the default value is used as an example).
- The 3PTY service mode: local 3PTY
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 7 7
huawei(config-if-sip-0)#mg-software parameter 28 0
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 three-party
enable
```

1.6.15.5 Configuring the Remote Three Party Conference Call Service

The remote three party conference call (3PTY) service enriches user experience and brings value-added services to carriers. The following describes the procedure for configuring the remote 3PTY service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In SIP mode, run the **mg-software parameter** command to set the 3PTY service mode.

Step 3 In SIP mode, run the **if-sip attribute optional** command to configure the conference factory URI of the SIP interface.

Step 4 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

----End

Example

To configure the remote 3PTY service for POTS user A with the listed data planning, run the following commands:

- The remote 3PTY service code of the POTS user is 21, and the service priority is 21 (the default value is used as an example).
- The 3PTY service mode: remote 3PTY
- The conference factory URI of the SIP interface: sip:alice@huawei.com.
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 21 21
huawei(config-if-sip-0)#mg-software parameter 28 1
huawei(config-if-sip-0)#if-sip attribute optional conference-factory-uri
sip:alice@huawei.com
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
```

To configure the remote 3PTY service for ISDN user A with the listed data planning, run the following commands:

- The remote 3PTY service code of the BRA user is 23, and the service priority is 23 (the default value is used as an example).
- The 3PTY service mode: remote 3PTY
- The conference factory URI of the SIP interface: sip:alice@huawei.com.
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 23 23
huawei(config-if-sip-0)#mg-software parameter 28 1
huawei(config-if-sip-0)#if-sip attribute optional conference-factory-uri
sip:alice@huawei.com
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
```

1.6.15.6 Configuring the Conferencing Service

The conferencing service enriches user experience and brings value-added services to carriers. This section describes how to configure the conferencing service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In SIP mode, run the **if-sip attribute optional** command to configure the conference factory URI of for the SIP interface.

Step 3 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 4 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the conferencing service permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the conferencing service permission for a service user.

----End

Example

To configure the conferencing service for POTS user A with the listed data planning, run the following commands:

- The conferencing service code of the POTS user is 0, and the service priority is 4 (the default value is used as an example).
- The conference factory URI of the SIP interface: sip:alice@huawei.com.
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 0 4
huawei(config-if-sip-0)#if-sip attribute optional conference-factory-uri
sip:alice@huawei.com
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 conf enable
```

To configure the conferencing service for ISDN user A with the listed data planning, run the following commands:

- The conferencing service code of the BRA user is 6, and the service priority is 6 (the default value is used as an example).
- The conference factory URI of the SIP interface: sip:alice@huawei.com.
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 6 6
huawei(config-if-sip-0)#if-sip attribute optional conference-factory-uri
sip:alice@huawei.com
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 conf enable
```

1.6.15.7 Configuring the Explicit Communication Transfer Service

The explicit communication transfer (ECT) service enriches user experience and brings value-added services to carriers. This section describes how to configure the ECT service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools

3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the ECT service permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the ECT service permission for a service user.

----End

Example

To configure the ECT service for POTS user A with the listed data planning, run the following commands:

- The ECT service code of the POTS user is 2, and the service priority is 3 (the default value is used as an example).
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 2 3
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 ct enable
```

To configure the ECT service for ISDN user A with the listed data planning, run the following commands:

- The ECT service code of the BRA user is 8, and the service priority is 8 (the default value is used as an example).
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 8 8
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 ct enable
```

1.6.15.8 Configuring the Call Forwarding Service

The call forwarding service enriches user experience and brings value-added services to carriers. This section describes how to configure the call forwarding service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority of a POTS user.

NOTE

- This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.
- The service priority of the ISDN user is configured on the softswitch.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 In the ESL mode, run the **sippstnuser servicedata parameter set** command to set the forwarded-to number of the POTS phone of the service user.

NOTE

The forwarded-to number of an ISDN user must be registered with the core network.

----End

Example

To configure the call forwarding service for POTS user A with the listed data planning, run the following commands:

- The call forwarding service code of the POTS user is 17, and the service priority is 17 (the default value is used as an example).
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.
- The call forwarding mode is Call Forwarding Busy (CFB) and the forwarded-to number is 12345678.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 17 17
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser servicedata parameter set 0/2/0 telno 83110000
cfbnum 12345678
```

To configure the call forwarding service for ISDN user A with the listed data planning, run the following commands:

- The conferencing service code of the BRA user is 18, and the service priority is 18 (the default value is used as an example).
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 18 18
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
```

1.6.15.9 Configuring the Emergency Call Service

The emergency call service enriches user experience and brings value-added services to carriers. This section describes how to configure the emergency call service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface

Procedure

Step 1 In SIP mode, run the **mg-software parameter (sip)** command to enable the emergency call permission on the interface.

Step 2 In config mode, run the **local-digitmap add** command to configure the emergency digitmap.

----End

Example

To configure the emergency call service on SIP interface 0 (name of the emergency call digitmap: huawei; digitmap body: 110|120|119), run the following commands:

```
huawei(config)#interface sip 0
hhuawei(config-if-sip-0)#mg-software parameter 6 1
huawei(config-if-sip-0)#quit
huawei(config)#local-digitmap add huawei emergency 110|120|119
```

1.6.15.10 Configuring the Hotline Service

The hotline service enriches user experience and brings value-added services to carriers. This section describes how to configure the hotline service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 For POTS users, run the **sippstnuser servicedata parameter set** command in ESL mode to set the hotline number for a service user. For ISDN users, run the **sipbrauser servicedata parameter set** command to set the hotline number for a service user.

Step 4 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the hotline service permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the hotline service permission for a service user.

----End

Example

To configure the hotline service for POTS user A with the listed data planning, run the following commands:

- The hotline service code of the POTS user is 12, and the service priority is 12 (the default value is used as an example).
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.
- The hotline number of user A is 83110001. In addition, set the hotline delay to 0s (immediate hotline mode).

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 12 12
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser servicedata parameter set 0/2/0 telno 83110000
```

```
hottime 0 hotlinenum 83110001
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 hotline enable
```

To configure the hotline service for ISDN user A with the listed data planning, run the following commands:

- The hotline service code of the BRA user is 13, and the service priority is 13 (the default value is used as an example).
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 8311010.
- The hotline number of user A is 83110001. The hotline delay to 5s (the default value is used as an example).

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 13 13
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser servicedata parameter set 0/6/0 telno 83110010
hottime 5 hotlinenum 83110011
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 hotline enable
```

1.6.15.11 Configuring the Message Waiting Indication Service

The Message Waiting Indication (MWI) service enriches user experience and brings value-added services to carriers. This section describes how to configure the MWI service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface

Procedure

Step 1 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 2 For POTS users, run the **sippstnuser rightflag set** command in ESL mode to set the MWI service permission for a service user. For ISDN users, run the **sipbrauser rightflag set** command to set the MWI service permission for a service user.

----End

Example

To configure the MWI service for POTS user A with the listed data planning, run the following commands:

- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 mwi enable
```

To configure the MWI service for ISDN user A with the listed data planning, run the following commands:

- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 mwi enable
```

1.6.15.12 Configuring the Connected Line Identification Presentation Service

The Connected Line Identification Presentation (COLP) service enriches user experience and brings value-added services to carriers. This section describes how to configure the COLP service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface

Procedure

Step 1 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 2 In SIP mode, run the **sipprofile modify** command to enable the COLP service on control point PF295.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 3 After the preceding configurations are complete, the service party subscribes to the COLP service on the phone.

----End

Example

To configure the COLP service for POTS user A with the listed data planning, run the following commands:

- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#quit
huawei(config)#interface sip 0
huawei(config-if-sip-0)#sipprofile modify syspara 295 1
```

To configure the COLP service for ISDN user A with the listed data planning, run the following commands:

- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#quit
huawei(config)#interface sip 0
huawei(config-if-sip-0)#sipprofile modify syspara 295 1
```

1.6.15.13 Configuring the Malicious Call Identification Service

The malicious call identification (MCID) service enriches user experience and brings value-added services to carriers. This topic describes how to configure the MCID service.

Prerequisites

The following configurations have been completed:

1. Configuring the Upstream VLAN Interface
2. Configuring the Media and Signaling IP Address Pools
3. Adding an SIP Interface
4. Configuring the Customized SIP Value-added Service Logic

NOTE

- If the preset service logic file cannot meet the requirements, you need to customize and load the customized service logic file. Otherwise, you do not need to customize the service logic file.
- If the customized service logic file exists, skip this step.

Procedure

Step 1 In SIP mode, run the **sipprofile modify** command to set the service priority.

NOTE

This command is used to control the program processing flow inside the system to resolve the adaptation and interworking issues for devices from different vendors. This command is used in the interworking test phase by Huawei engineers. After the interworking test is passed, modification of parameters of this command is not needed in most cases. Incorrect values of parameters of this command may lead to interworking failures. Do not change parameters of this command by yourself. If you need to change parameter values, contact Huawei technical support.

Step 2 In ESL mode, run the **sippstnuser add** command to add an SIP PSTN user or run the **sipbrauser add** command to add an SIP ISDN user.

Step 3 For POTS users, run the **sippstnuser permissionflag set** command in ESL mode to set the MCID service permission for a service user. For ISDN users, run the **sipbrauser permissionflag set** command to set the MCID service permission for a service user.

----End

Example

To configure the MCID service for POTS user A with the listed data planning, run the following commands:

- The MCID service code of the POTS user is 4, and the service priority is 0 (the default value is used as an example).
- The ASPB service board is in slot 0/2. The PSTN user A is added to port 0/2/0 on SIP interface 0.
- The phone number of user A is 83110000.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 4 0
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/0 0 telno 83110000
huawei(config-esl-user)#sippstnuser rightflag set 0/2/0 telno 83110000 mcid enable
```

To configure the MCID service for ISDN user A with the listed data planning, run the following commands:

- The conferencing service code of the ISDN user is 10, and the service priority is 10 (the default value is used as an example).
- The DSRD service board is in slot 0/6. The BRA user A is added to port 0/6/0 on SIP interface 0.
- The phone number of user A is 83110010.

```
huawei(config-if-sip-0)#sipprofile modify srv-pri 10 10
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipbrauser add 0/6/0 0 telno 83110010
huawei(config-esl-user)#sipbrauser rightflag set 0/6/0 telno 83110010 mcid enable
```

1.7 MGCP Voice Feature

This topic describes the MGCP protocol and the working principle of MGCP application in VoIP, MoIP and FoIP.

1.7.1 Introduction to the MGCP Feature

Definition

Defined by IETF, MGCP is a protocol that specifies a call control mechanism in which call control and service bearing are separated. Call control is independent of the media gateway (MG) and is processed by the MGC. Therefore, MGCP is actually a master-slave protocol. The MG establishes various service connections under the control of the MGC.

MGCP provides the following commands:

1. NotificationRequest: The MGC sends this command to request the MG to detect a specified event, such as an offhook event or onhook event. After detecting such an event, the MG notifies the MGC. Through this command, the MGC can also instruct the MG to play signal tones, such as the dial tone and busy tone.
2. Notify: After the MG detects the specified event as instructed by the MGC, the MG sends this command to notify the MGC of the detected event.
3. CreateConnection: The MGC sends this command to instruct the MG to create a media connection. The command contains the instruction or suggestion on the bearing parameters and connection parameters.
4. ModifyConnection: The MGC sends this command to instruct the MG to modify the bearing parameters and connection parameters of an established media connection.
5. DeleteConnection: The MGC sends this command to instruct the MG to delete an established media connection. The MG can also voluntarily delete a connection. This means that, when the MG discovers that system resources are insufficient or the system is faulty, the MG can delete the connection and at the same time send this command to notify the MGC. Therefore, this command is bi-directionally available between the MGC and the MG.
6. AuditEndpoint and AuditConnection: The MGC sends the commands to check the status of a specified endpoint and connection.
7. RestartInProgress: The MG sends this command to notify the MGC that the MG or a certain endpoint managed by the MG is not available or is becoming available. This command is usually triggered by a system fault or restart.

MGCP also provides the following features:

1. Encoding in the text format
2. Adopting the Session Description Protocol (SDP) to describe the connection parameters of the media stream
3. Introducing the concept of event package
4. Adopting the wildcard to describe endpoints and events

Purpose

MGCP solves the internal problems of MG and media devices, thus realizing an open distributed system which is formed by the MG and media devices.

In the MGCP mechanism, the MG and media devices are separated into two logically independent parties, the MG and the MGC, which communicate through MGCP. The MG processes the user plane, and the MGC processes the control plane and controls the actions of the MG. In other words, the MG acts under the control of the MGC.

1.7.2 MGCP Principles

1.7.2.1 MGCP-Based VoIP

Figure 1-89 illustrates the principle of the call establishment and release in the MGCP-based VoIP service.

Figure 1-89 Principles of the call establishment and release in the MGCP-based VoIP service

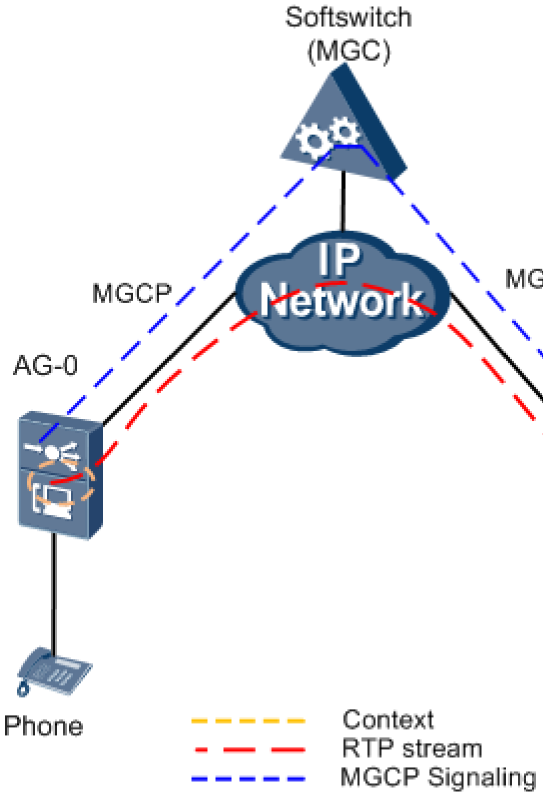
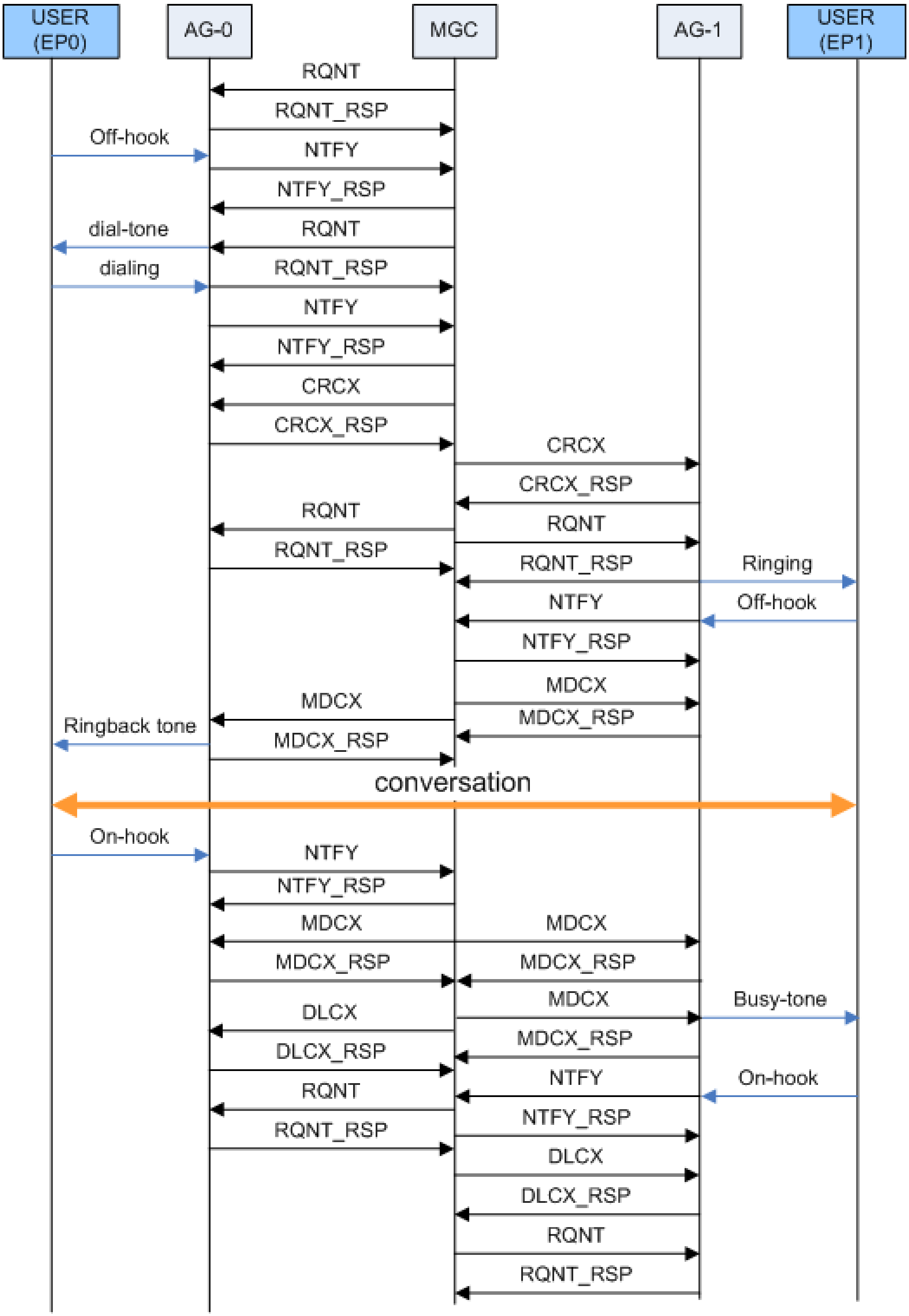


Figure 1-90 illustrates the basic flow of a call establishment and release process.

Figure 1-90 MGCP-based call flow



1. AG-0 detects the offhook of EP0, and notifies the MGC of the offhook event through the Notify command.

2. After the MGC receives the offhook event, the MGC sends a digitmap to AG-0, requests AG-0 to play the dial tone to EP0, and at the same time checks for the digit collection event.
3. User EP0 dials a telephone number, and AG-0 collects the digits according to the digitmap issued by the MGC. Then, AG-0 reports the result of digit collection to the MGC.
4. The MGC sends the CRCX (CreateConnection) command to AG-0 requesting AG-0 to create a connection at endpoint EP0.
5. AG-0 allocates resources for creating this connection and sends a response to the MGC. The response contains the session description that provides the necessary information for the peer end to send the packet to AG-0, such as the IP address and UDP port number.
6. The MGC sends the CRCX command to AG-1 requesting AG-1 to create a connection at endpoint EP1.
7. AG-1 allocates resources for creating this connection and sends a response to the MGC. The response contains the session description that provides the necessary information for the peer end to send the packet to AG-1, such as the IP address and UDP port number.
8. AG-1 detects the offhook of EP1, and sends the Notify command to the MGC. The softswitch (MGC) sends the MDCX (ModifyConnection) command to stop the ring back tone of EP0 and the ringing of EP1.
9. The MGC sends the session description of AG-1 to EP0 through the MDCX command. Then, the conversation is set up between EP0 and EP1.
10. AG-0 detects the onhook of EP0, and notifies the MGC of the onhook event through the Notify command.
11. The MGC sends the MDCX command to AG-0 and AG-1 respectively to modify the RTP resource to receive-only.
12. The MGC sends the MDCX command to AG-1 requesting AG-1 to play the busy tone to EP1, and at the same time checks for the onhook event.
13. The MGC sends the DCLX (DeleteConnection) command to AG-0, requesting AG-0 to release the resources that are occupied by the call of EP0.
14. AG-1 detects the onhook of EP1, and notifies the MGC of the onhook event through the Notify command.
15. The MGC sends the DCLX command to AG-1, requesting AG-1 to release the resources that are occupied by the call of EP1.
16. The call between EP0 and EP1 is terminated, and all the resources occupied by the call are released.

1.7.2.2 MGCP-Based MoIP

MoIP refers to the modem service provided on the IP network or between the IP network and the traditional PSTN network. According to different control devices, MoIP can be classified as softswitch-controlled MoIP and auto-switching MoIP.

Softswitch-Controlled MoIP

The basic flow of the softswitch-controlled MoIP service is as follows:

1. Establish a call. If the MoIP service is configured on the softswitch, the softswitch sends a command to the MG instructing the MG to detect the modem event.
2. The calling party and called party start communicating with each other.

3. During the call, when the MG detects the ANS or ANSAM modem start event (low-speed modem signal), or detects the ANSBAR or ANSAMBAR modem start event (high-speed modem signal), the MG sends the event to the softswitch.
4. According to the event, the softswitch sends a command instructing the MG to switch the DSP channel of the calling and called parties to the low-speed or high-speed modem mode.
5. According to the command sent by the softswitch, the MG switches the DSP channel to the corresponding modem mode. At this stage, the MG adopts the encoding format and port number specified by the softswitch.
6. The settings of echo cancellation (EC), voice activity detection (VAD), and DSP working mode are as follows:
   a. Low-speed modem: EC-ON, VAD-OFF, DSP working mode-modem mode
   b. High-speed modem: EC-OFF, VAD-OFF, DSP working mode-modem mode
7. After the modem data is transmitted, if the conversation proceeds, the DSP working mode does not automatically switch from the modem mode to the voice mode, because the modem end event is not issued. As a result, the quality of the voice service may be affected.

Auto-Switching MoIP

The basic flow of the auto-switching MoIP service is as follows:

1. Set up a conversation.
2. The MGs at both ends check for the modem event on the IP side and the TDM side. When the modem event is detected, if the modem transmission mode is configured as auto-switching, the coding mode is switched to G.711 (the a/μ law is configurable), and the DSP parameters are modified according to the modem mode (high-speed/low-speed) detected.
3. When the modem service is terminated, the call is released.

1.7.2.3 MGCP-Based FoIP

FoIP refers to the fax service provided on the IP network or between the IP network and the traditional PSTN network. The fax machine can be regarded as a special modem. In the FoIP negotiation, the modem negotiation is performed before the fax negotiation.

According to the transmission protocol adopted, there are two modes of fax services carried on the IP network: the T.30 transparent transmission mode and the T.38 mode. According to different control devices, FoIP can be classified as softswitch-controlled FoIP and auto-switching FoIP.

Softswitch-Controlled FoIP

The fax service can be classified into high-speed fax and low-speed fax. The softswitch-controlled low-speed fax service supports the T.30 transparent transmission mode and the T.38 mode. The basic service flow is as follows:

1. Configure the fax service and fax flow on the MGs and the softswitch.
2. After the voice channel is set up, the softswitch instructs the MG to detect the fax event and modem event.
3. When detecting the fax event, the MG reports the event to the softswitch. The event can be a low-speed modem event (ANS or ANSAM) or a low-speed fax event (V.21Flag).

4. According to the preset fax flow, the softswitch instructs the MGs at both ends to change the DSP channel working mode to the T.30 transparent transmission mode or T.38 mode.
5. The fax starts.
6. After the fax is complete, if the MG detects the fax end event, the MG reports the event to the softswitch.
7. The softswitch instructs the MGs at both ends to change the DSP channel working mode to the voice mode.
8. The voice service proceeds.

The softswitch-controlled high-speed fax service supports the T.30 transparent transmission mode. The basic service flow is as follows:

1. Configure the fax service and fax flow on the MGs and the softswitch.
2. After the voice channel is set up, the softswitch instructs the MG to detect the fax event and modem event.
3. When detecting a fax event, the MG reports the event to the softswitch. The event can be a high-speed modem event (ANSBAR or ANSAMBAR) or a low-speed fax event (V.21Flag; if the peer end is a low-speed fax machine or the network quality is poor, the fax speed is automatically decreased and this event is reported).
4. According to the preset fax flow, the softswitch instructs the MGs at both ends to change the DSP channel working mode to T.30 transparent transmission mode.
5. The fax starts.
6. After the fax is complete, if the MG detects the fax end event, the MG reports the event to the softswitch.
7. The softswitch instructs the MGs at both ends to change the DSP channel working mode to the voice mode. The voice service proceeds.

Auto-Switching FoIP

The auto-switching fax service supports the T.30 transparent transmission mode and the T.38 mode. The basic service flow is as follows:

1. Configure the auto-switching fax service on the MGs at both ends.
2. Establish a call and use the voice service.
3. The MG checks for the fax event on the IP side and the TDM side. When detecting the fax event, the MG changes the DSP channel working mode to the T.30 transparent transmission mode or the T.38 mode.
4. After the fax is complete, when the MG detects the fax end event, the MG changes the DSP channel working mode to the voice mode.
5. The voice service proceeds.

Common Fax Protocols

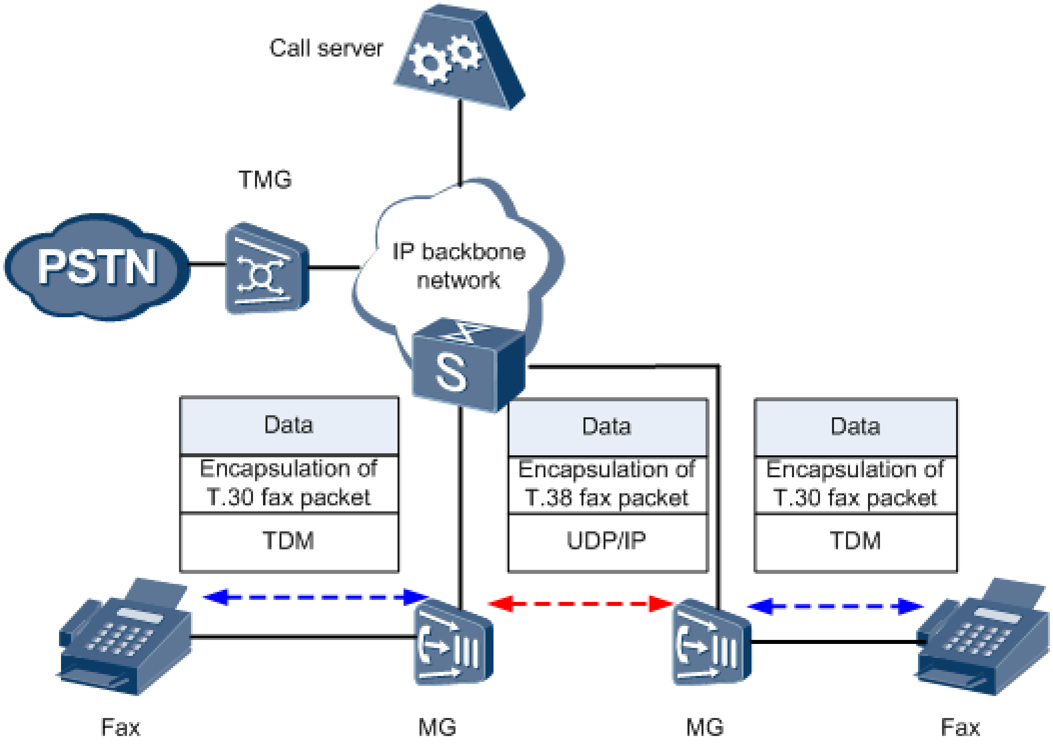
Two protocols are usually used for implementing the fax service on the packet voice network: the ITU-T Recommendation T.30 and ITU-T Recommendation T.38.

T.30 is based on the PSTN network. T.30 particularly defines the flow for transmitting fax signals on the PSTN network. It also defines the modulation mode (V.17/V.21/V.27/V.29/V.34) and transmission format (HDLC) of data, and the physical standard for fax signals. The T.30 fax messages and data can be transmitted transparently between MGs. This is called the T.30

transparent transmission mode. The quality of the fax in this mode may not be high due to packet loss, latency, and disorder on the IP network.

T.38 is a real-time fax mode based on the IP network. In this mode, the MG terminates the T.30 signals sent from the fax machine, and transmits the data to the peer MG in the T.38 mode. The peer MG then receives the T.38 packets and converts the packets into T.30 signals. The merit of the T.38 fax is that the fax packets have a redundancy processing mechanism and do not strictly rely on the quality of the network (the fax service can be processed even when a 20% packet loss occurs on the network). The demerit is that the DSP chip needs to participate in parsing the T.30 signals. Because there are various types of terminals on the network, the compatibility problem may arise.Figure 1-91 illustrates the principle of the T.38 fax.

Figure 1-91 Principle of the T.38 fax


Call server
TMG
PSTN
IP backbone network
S
Data
Encapsulation of T.30 fax packet
TDM
Data
Encapsulation of T.38 fax packet
UDP/IP
Data
Encapsulation of T.30 fax packet
TDM
Fax
MG
MG
Fax

1.7.3 MGCP Standards and Protocols Compliance

- RFC2705
- RFC3405
- T.30: It is based on the PSTN network. T.30 particularly defines the flow for transmitting fax signals on the PSTN network. It also defines the modulation mode (V.17/V.21/V.27/V.29/V.34) and transmission format (HDLC) of data, and the physical standard for fax signals. The T.30 fax messages and data can be transmitted transparently between MGs. This is called the T.30 transparent transmission mode. The quality of the fax in this mode may not be high due to packet loss, latency, and disorder on the IP network.
- T.38: It is a real-time fax mode based on the IP network. In this mode, the MG terminates the T.30 signals sent from the fax machine, and transmits the data to the peer MG in the T.38 mode. The peer MG then receives the T.38 packets and converts the packets into

T.30 signals. The merit of the T.38 fax is that the fax packets have a redundancy processing mechanism and do not strictly rely on the quality of the network (the fax service can be processed even when a 20% packet loss occurs on the network). The demerit is that the DSP chip needs to participate in parsing the T.30 signals. Because there are various types of terminals on the network, the compatibility problem may arise.

1.8 H.248 Voice Feature

This topic first describes the H.248 protocol, and then describes the protocol mechanism, and last describes the application of H.248 in VoIP, MoIP, and FoIP.

1.8.1 Introduction to the H.248 Feature

Definition

H.248 is a media gateway control protocol through which the media gateway controller (MGC) controls the media gateway (MG) so that interoperability is implemented between different media. ITU-T issued the first version of this protocol in June 2000.

Purpose

Compared with MGCP, H.248 has the following merits:

- Supports more types of access technologies, and is more thorough and complete in standardization
- Compensates for the deficiency of MGCP in descriptiveness, is applicable to larger networks and has better extensibility and flexibility
- Carried on various protocols, such as UDP/SCTP (MGCP is carried on UDP)

1.8.2 H.248 Principles

1.8.2.1 Mechanism of the H.248 Protocol

Termination ID

A termination ID identifies a termination that is going to register or deregister a service. The termination ID of each termination is unique. During service configuration, the termination ID corresponding to each termination must be configured on the MG and the MGC. The root termination ID represents an entire MG. The ServiceChange command executed on the root termination ID is effective on an entire MG. The wildcarding principle is that the ALL wildcard (*) can be used but the CHOOSE wildcard ($) cannot be used.

Registration Mechanism of the H.248 Interface

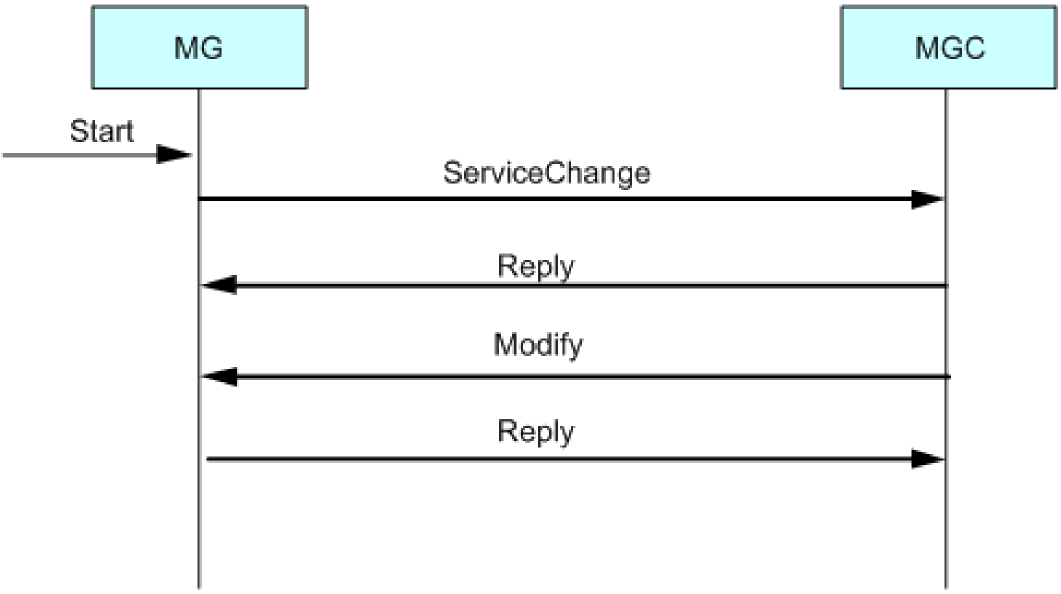
The MG sends the ServiceChangeRequest command to inform the MGC that a user or a group of users are about to register or deregister service. After this command is executed successfully, the termination status is changed to InService or OutOfService. In addition, the MGC can unsolicitedly send the ServiceChangeRequest command to request the MG to register or deregister service for a user or a group of users.

NOTE

Currently, the MG does not support the MGC to unsolicitedly send the ServiceChangeRequest command requesting the MG to register service for a user or a group of users.

Figure 1-92 shows the registration flow of the MG.

Figure 1-92 Registration flow of the MG



Description of the flow:

1. The MG sends the ServiceChangeRequest command to the MGC. In the command, TerminationId is Root, Method is Restart, and ServiceChangeReason is 901 (cold boot, registering for the first time after power-on), 902 (warm boot, through command lines), or 900 (in other cases).
2. The MGC sends the Reply message to the MG indicating the successful registration.
3. The MGC sends the Modify command to the MG requesting the MG to detect the offhook of all users (al/of).
4. The MG responds to the MGC with the Reply message.

Heartbeat Mechanism of the H.248 Interface

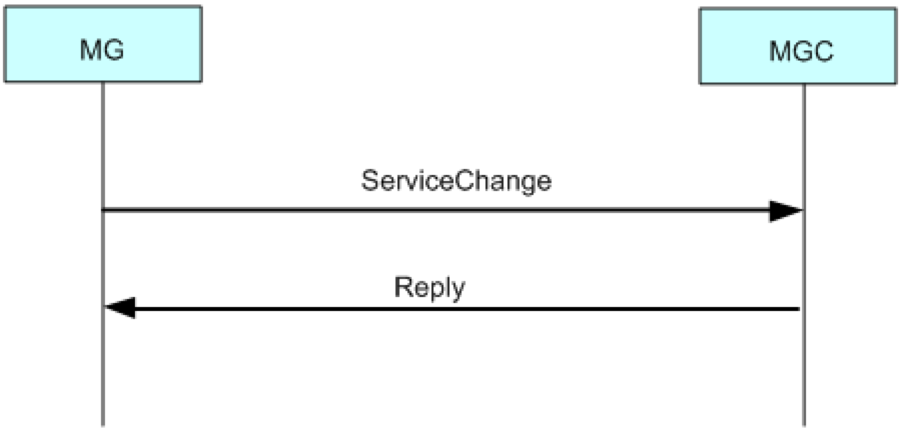
After the registration is successful, the MG and the MGC maintain communication by sending each other the heartbeat message Notify (it/ito). By default, the heartbeat message is sent every 60s. The sending interval can be set within the range of 5-655s.

After the MG sends the first heartbeat message to the MGC, if the MG does not receive the heartbeat response from the MGC before the preset interface heartbeat timer (for example, the length of three sending intervals) times out, the MG sets the interface status to "wait for response". Then, the MG keeps initiating a registration with the MGC. If dual-homing is configured, the MG initiates registration with the two MGCs alternatively. The registration is initiated once every 30s, every three trials of registration are one round, and every registration message is re-transmitted 7 times. Therefore, 24 registration messages in total are transmitted within 90s. Then, the MG starts the next round of registration with the other MGC.

Deregistration Mechanism of the H.248 Interface

Figure 1-93 shows the unsolicited deregistration flow of the MG.

Figure 1-93 Unsolicited deregistration flow of the MG

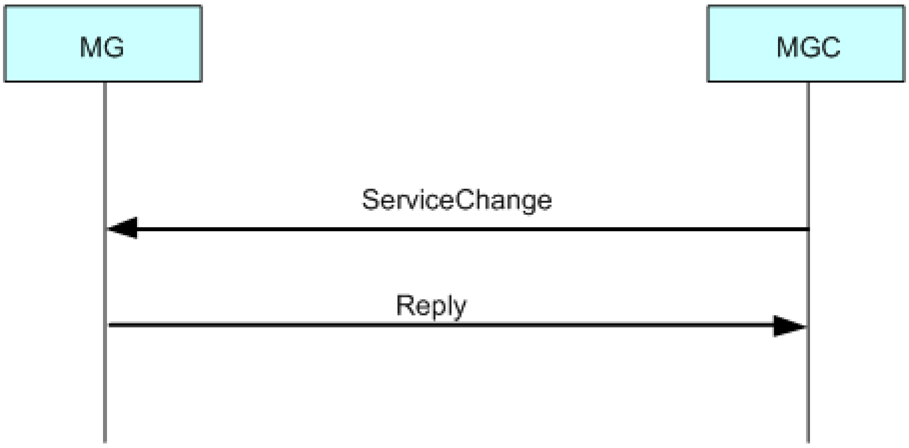


Description of the flow:

1. The MG sends the ServiceChangeRequest command to the MGC. In the command, TerminationId is Root, Method is Forced, and ServiceChangeReason is 905 ("905" indicates that the termination is taken out of service because of maintenance operation, and now the MG uses "905" to initiate a deregistration request through command lines).
2. The MGC sends the Reply message to the MG indicating a successful deregistration.

Figure 1-94 shows the flow of the MGC unsolicitedly deregistering the MG.

Figure 1-94 Unsolicited deregistration flow of the MGC



Description of the flow:

1. The MGC sends the ServiceChangeRequest command to the MG. In the command, TerminationId is Root, Method is Forced, and ServiceChangeReason is 905.
2. The MG responds to the MGC with the Reply message. The access device (MG) supports the registration and deregistration of not only an entire MG but also a single

termination. The service status of a single user can be changed through the registration and deregistration of a single termination.

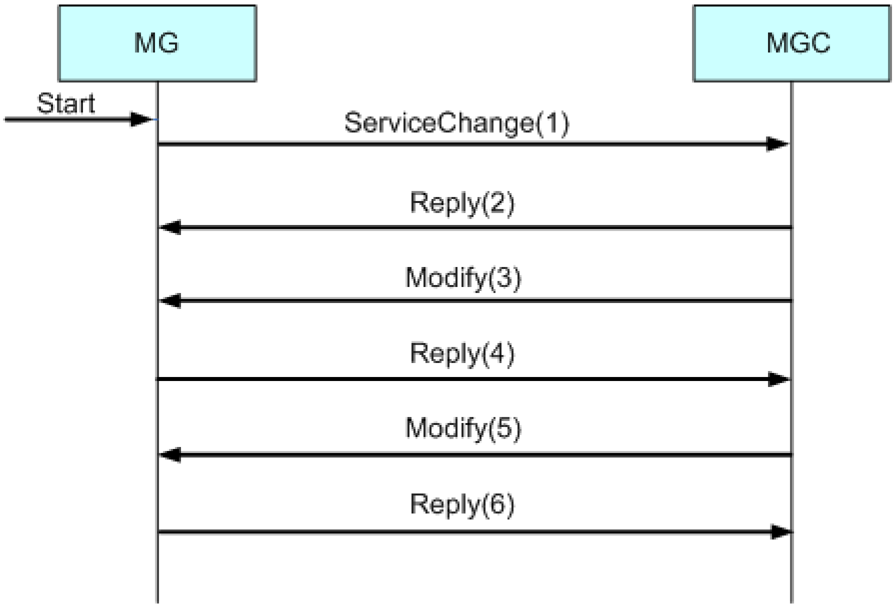
Authentication Mechanism of the H.248 Interface

Authentication is a security mechanism through which the MGC authenticates the legality of the MG user. The purpose of authentication is to prevent unauthorized entities from establishing illegal calls or interfering with legal calls through the H.248 or MGCP protocol. Authentication can be implemented only when it is also supported by the softswitch interconnected with the MG.

- In H.248, the implementation of authentication complies with RFC2402.
- MD5 is adopted as the encryption algorithm.

Figure 1-95 shows the authentication flow.

Figure 1-95 Authentication flow



The basic flow is as follows:

1. The MG sends the ServiceChange command to register with the MGC. The command contains the digital signature of the MG.
2. After receiving the ServiceChange command, the softswitch verifies the MG and sends a reply.
3. The softswitch sends the Modify message to the MG. The message contains the required algorithm ID and random number.
4. The MG verifies the message sent by the softswitch and sends a reply.
5. The softswitch authenticates the MG periodically.
6. The MG sends replies to the softswitch.

1.8.2.2 H.248-Based VoIP

Figure 1-96 shows the principles of call establishment and release in the H.248-based VoIP service.

Figure 1-96 Principles of the H.248-based VoIP service

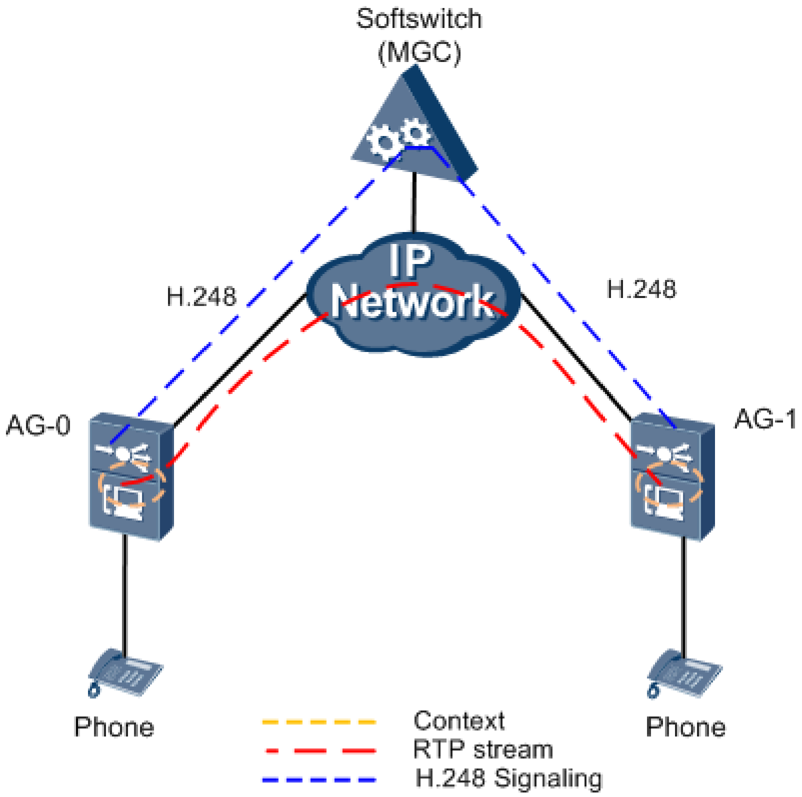
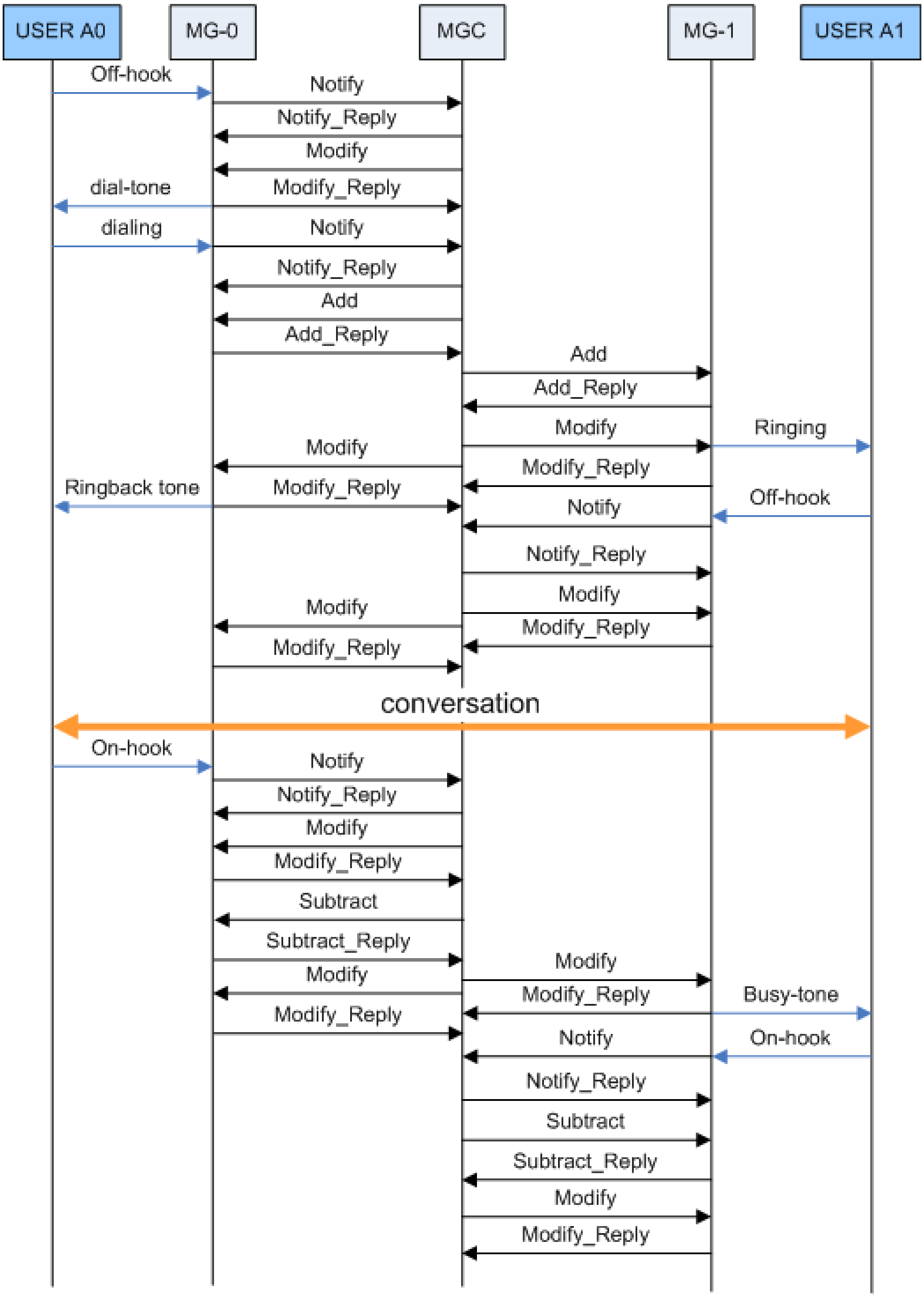


Figure 1-97 shows the flowchart of call establishment and release in the H.248-based VoIP service.

Figure 1-97 Flowchart of call establishment and release in the H.248-based VoIP service



1. MG-0 detects the offhook of user A0, and notifies the MGC of the offhook event through the Notify command.
2. After receiving the offhook event, the MGC sends a digitmap to MG-0, requests MG-0 to play the dial tone to user A0, and at the same time checks for the digit collection event.

3. User A0 dials a telephone number, and MG-0 collects the digits according to the digitmap issued by the MGC. Then, MG-0 reports the result of digit collection to the MGC.
4. The MGC sends the Add command to MG-0 for creating a context and adding the termination and RTP termination of user A0 into the context.
5. After creating the context, MG-0 responds to the MGC. The response contains the session description that provides the necessary information for the peer end to send the packet to MG-0, such as the IP address and UDP port number.
6. The MGC sends the Add command to MG-1 for creating a context and adding the termination and RTP termination of user A1 into the context, and then issues the IP address/UDP port ID of user A0 to user A1.
7. After creating the context, MG-1 responds to the MGC. The response contains the session description that provides the necessary information for the peer end to send the packet to MG-1, such as the IP address and UDP port ID.
8. MG-1 detects the offhook of user A1, and then reports the offhook event to the MGC. The softswitch (MGC) sends the Modify command to stop the ring back tone of user A0 and the ringing of user A1.
9. The MGC sends the session description of MG-1 to user A0 through the Modify command. Then, the conversation is set up between users A0 and A1.
10. MG-0 detects the onhook of user A0, and notifies the MGC of the onhook event through the Notify command.
11. The MGC sends the Modify command to MG-0 and MG-1 respectively to modify the RTP mode to receive-only.
12. The MGC sends the Modify command to MG-1 requesting MG-1 to play the busy tone to user A1, and at the same time checks for the onhook event.
13. The MGC sends the Subtract command to MG-0, requesting MG-0 to release the resources that are occupied by the call of user A0.
14. MG-1 detects the onhook of user A1, and notifies the MGC of the onhook event through the Notify command.
15. The MGC sends the Subtract command to MG-1, requesting MG-1 to release the resources that are occupied by the call of user A1.
16. The call between users A0 and A1 is terminated, and all the resources occupied by the call are released.

1.8.2.3 H.248-Based MoIP

This topic describes the principle of the H.248-based modem over Internet protocol (MoIP) service.

MoIP refers to the modem service provided on the IP network or between the IP network and the traditional PSTN network. According to different control devices, MoIP can be classified as softswitch-controlled MoIP and self-switching MoIP.

Softswitch-Controlled MoIP

The basic flow of the softswitch-controlled MoIP service is as follows:

1. Establish a call. If the MoIP service is configured on the softswitch, the softswitch sends a command to the MG instructing the MG to detect the modem event.
2. The calling party and called party start communicating with each other.

3. During the call, when the MG detects the ANS or ANSAM modem start event (low-speed modem signal), or detects the ANSBAR or ANSAMBAR modem start event (high-speed modem signal), the MG sends the event to the softswitch.
4. According to the event, the softswitch sends a command instructing the MG to switch the DSP channel of the calling and called parties to the low-speed or high-speed modem mode.
5. According to the command sent by the softswitch, the MG switches the DSP channel to the corresponding modem mode. At this stage, the MG adopts the encoding format and port number specified by the softswitch.
6. The settings of echo cancellation (EC), voice activity detection (VAD), and DSP working mode are as follows:
   a. Low-speed modem: EC-ON, VAD-OFF, DSP working mode-modem mode
   b. High-speed modem: EC-OFF, VAD-OFF, DSP working mode-modem mode
7. After the modem data is transmitted, if the conversation proceeds, the DSP working mode does not automatically switch from the modem mode to the voice mode, because the modem end event is not issued.

Self-switching MoIP

The basic flow of the self-switching MoIP service is as follows:

1. Set up a conversation.
2. The MGs at both ends check for the modem event on the IP side and the TDM side. When the modem event is detected, if the modem transmission mode is configured as self-switching, the coding mode is switched to G.711 (the A/μ law is configurable), and the DSP parameters are modified according to the modem mode (high-speed/low-speed) detected.
3. When the modem service is terminated, the call is released.

1.8.2.4 H.248-Based FoIP

Fax over IP (FoIP) is used to provide fax services on an IP network or between an IP network and a traditional public switched telephone network (PSTN) network. A fax machine can be regarded as a special modem. In FoIP negotiation, modem negotiation is performed before fax negotiation.

According to different transmission protocols, two modes are available for transmitting fax services on the IP network: T.30 transparent transmission mode and T.38 transmission mode. FoIP are classified into softswitch-controlled FoIP and self-switching FoIP according to different control devices.

Common Concepts for Fax Services

- Fax parameter negotiation mode
  - Parameters involved in negotiation: codec mode, fax mode, IP port ID, voice activity detection (VAD), packetization interval, and echo cancellation (EC).
  - Self-switch:

    After a media gateway (MG) detects a fax signal tone, it automatically uses a transmission mode according to system configurations and does not need to send a signaling to the peer end. The transmission mode can be transparent transmission or T.38-based transmission.
  - Negotiation:

After an MG detects a fax signal tone, it negotiates the fax parameters with the peer end.

- Negotiation process
  - V2 negotiation process:

    The process focuses on the MG capability. The MG determines the transmission mode to be used (transparent transmission or T.38-based transmission), and determines whether the fax port ID is increased by 2. No signaling negotiation is performed when the port ID is increased by 2. Therefore, when the V2 T.38 is configured, the IDs of the local and peer ports must be configured consistently. That is, the port IDs must be increased by 2 or not increased by 2 at the same time.

  - V3 negotiation process:

    The process focuses on the softswitch capability. If the softswitch requires the T.38-based transmission, this transmission mode is used; otherwise, the transparent transmission mode is used. The softswitch uses the signaling negotiation to determine whether the fax port ID is increased by 2.

NOTE

V2 and V3 indicate the negotiation process versions and are both defined by China Telecom.

Softswitch-Controlled FoIP

A fax service can be classified into high-speed fax and low-speed fax. The softswitch-controlled low-speed fax service supports the T.30 transparent transmission mode or the T.38 transmission mode. The basic service flow is as follows:

1. A user configures the fax service and fax flow on MGs and a softswitch.
2. After a voice channel is set up, the softswitch instructs an MG to detect a fax or modem event.
3. When detecting a fax event, the MG reports the event to the softswitch. The event can be a low-speed modem event (ANS or ANSAM) or a low-speed fax event (V.21Flag).
4. According to the preset fax flow, the softswitch instructs the MGs at both ends to change the digital signal processing (DSP) channel working mode to the T.30 transparent transmission mode or T.38 transmission mode.
5. The fax service starts.
6. After the fax service is complete, the MG reports the event to the softswitch if the MG detects a fax end event.
7. The softswitch instructs the MGs at both ends to change the DSP channel working mode back to the voice mode.
8. The voice service proceeds.

The softswitch-controlled high-speed fax service supports the T.30 transparent transmission mode. The basic service flow is as follows:

1. A user configures the fax service and fax flow on MGs and a softswitch.
2. After a voice channel is set up, the softswitch instructs an MG to detect a fax or modem event.
3. When detecting a fax event, the MG reports the event to the softswitch. The event can be a high-speed modem event (ANSBAR or ANSAMBAR) or a high-speed fax event (V.21Flag; if the peer end is a low-speed fax machine or network quality is poor, the fax speed is automatically decreased and this event is reported).

4. According to the preset fax flow, the softswitch instructs the MGs at both ends to change the DSP channel working mode to T.30 transparent transmission mode.
5. The fax service starts.
6. After the fax service is complete, the MG reports the event to the softswitch if the MG detects a fax end event.
7. The softswitch instructs the MGs at both ends to change the DSP channel working mode back to the voice mode.
8. The voice service proceeds.

Self-switching FoIP

The self-switching fax service supports the T.30 transparent transmission mode and the T.38 transmission mode. The basic service flow is as follows:

1. A user configures the self-switching fax service on MGs at both ends.
2. After a call connection is set up, the voice service starts.
3. The MG checks for a fax event on the IP side and the TDM side. When detecting a fax event, the MG changes the DSP channel working mode to the T.30 transparent transmission mode or the T.38 transmission mode.
4. After the fax service is complete, the MG changes the DSP channel working mode to the voice mode if the MG detects a fax end event.
5. The voice service proceeds.

Transparent Transmission Mode (T.30-Based Transmission Mode)

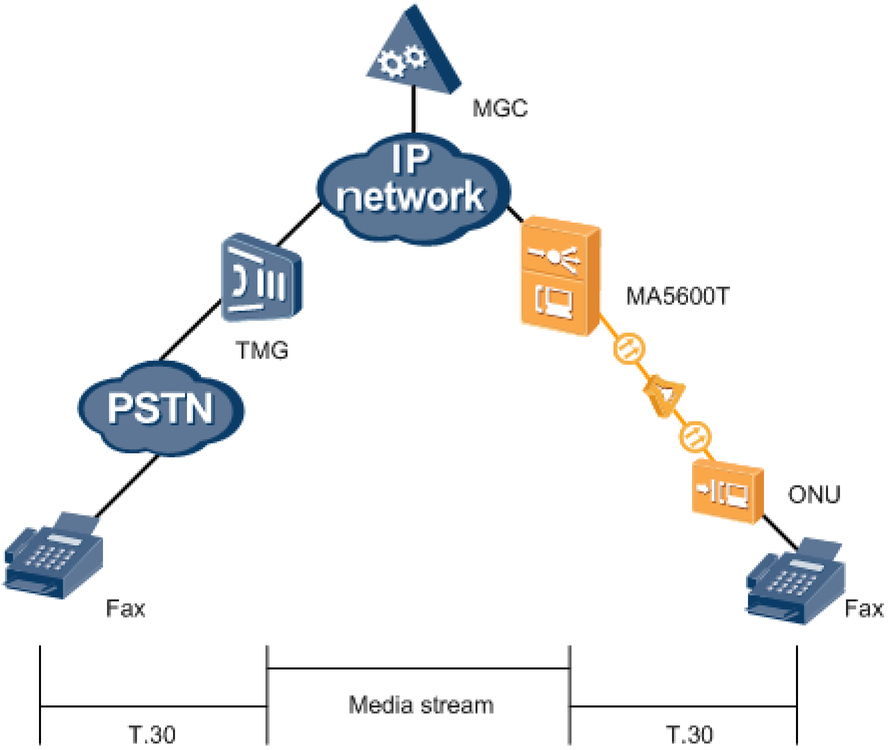
ITU-T Recommendation T.30 (T.30) is a PSTN-based fax protocol. The protocol defines the transmission flow, data modulation mode, data transmission format, and physical standards for fax signals on the PSTN. Transparent transmission enables T.30 fax messages and data to be transmitted transparently inside MGs or between MGs.

Transparent transmission has the following advantages and disadvantages:

- Advantages: Less DSP resource is consumed and there is less dependence on the softswitch.
- Disadvantages: Anti-interference capability is weak and fax quality is poor.

Figure 1-98 shows the transparent transmission mode.

Figure 1-98 Transparent transmission mode



T.38-Based Transmission Mode

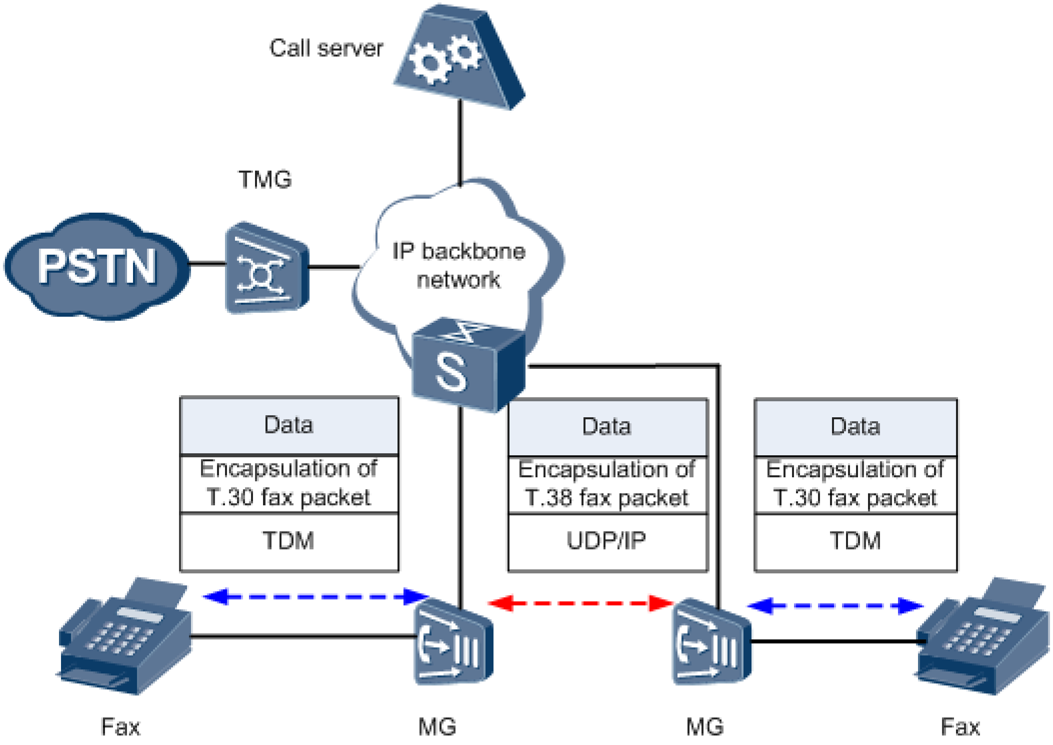
ITU-T Recommendation T.38 (T.38) is an IP-based real-time fax mode. In this mode, an MG terminates T.30 signals sent from a fax machine, and transmits the data to a peer MG in T.38 mode. The peer MG receives T.38 packets and converts them to T.30 signals.

The transmission mode has the following advantages and disadvantages:

- Advantages: A redundancy processing mechanism is provided and there is no strict requirements on network quality (the fax service can be successful even when a 20% packet loss occurs on the network).
- Disadvantages: The DSP on the MG needs to participate in parsing the T.30 signals. Compatibility problems may occur due to various terminal types.

Figure 1-99 shows the T.38-based transmission mode.

Figure 1-99 Principles of the T.38-based transmission mode


Call server
TMG
PSTN
IP backbone network
S
Data
Encapsulation of T.30 fax packet
TDM
Data
Encapsulation of T.38 fax packet
UDP/IP
Data
Encapsulation of T.30 fax packet
TDM
Fax
MG
MG
Fax

Comparison Between the High-Speed and Low-Speed Faxes

The high-speed fax differs from the low-speed fax in the following aspects:

- Standard: The high-speed fax uses V.8 data transmission process. The low-speed fax uses the transmission process defined by T.30. Some low-speed fax terminals may use earlier standards.
- Rate range: The rate range is 2400-33600 bit/s for the high-speed fax and 2400-14400 bit/s for the low-speed fax.
- Upstream transmission mode: The high-speed fax uses only the transparent transmission mode. The mode is high-speed modem transmission for the MG. The low-speed fax uses the transparent transmission mode or the T.38 transmission mode based on configurations.
- Requirement for the error correction mode: The error correction mode is mandatory for the high-speed fax and is optional for the low-speed fax.
- EC requirement: The high-speed fax provides an EC mechanism by itself and therefore the EC must be disabled. The low-speed fax requires that the EC be enabled.

1.8.3 H.248 Standards and Protocols Compliance

RFC3525

1.9 POTS Access

This topic describes the features in relation to the POTS interface, including basic features such as ringing and Z interface and enhanced features.

1.9.1 Introduction to POTS Access

Definition

Plain Old Telephone Service (POTS) is the traditional basic telephony service provided using twisted-pair copper lines. POTS Voice interface features are the features implemented on the voice interface.

Purpose

The purpose is to provide the standard-compliant voice interface that has the reliable protection capability and intelligent energy-saving function.

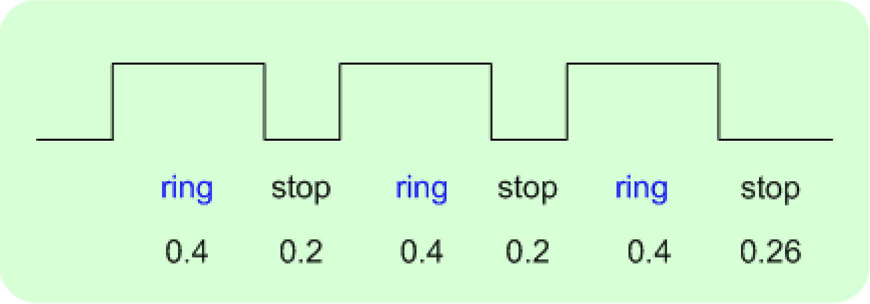
1.9.2 Ringing

A subscriber's phone rings, indicating that an incoming call is waiting on the line. The POTS board plays the ring tone to plain old telephone service (POTS) subscribers. This section describes the ring tone played by the POTS board.

Break-Make Ratio of the Ring Tone Played by the POTS Board

The break-make ratio varies depending on countries. For example, the break-make ratio in China is 1:4, that is, the ring tone is played in the cycle of one-second ringing followed by four-second mute. Figure 1-100 shows a six-segment ring with break-make ratio 0.4:0.2:0.4:0.2:0.4:2.6, which cycles in the following mode: ringing for 0.4 seconds and mute for 0.2 seconds twice followed by 0.4-second ringing and 2.6-second mute.

Figure 1-100 Ring example



Application

In some countries, break-make ratio is different for different services. For example, break-make ratio 1:4 is used for local calls, 1:2 for toll calls, and 0.4:0.2:0.4:4 for intra-Centrex calls.

1.9.3 POTS Interface Protection

Service boards are connected to user terminals through subscriber cables. Some subscriber cables may be routed under the ground or overhead and some may be routed in parallel with the mains AC power cables. In these cases, a high voltage may be generated on subscriber cables because of the lightning attack, contact with power lines, and induction of power lines. The high voltage may damage the ports on service boards. Therefore, service boards must be equipped with the protection capability to prevent occurrence of the preceding problems.

1.9.4 Features of the POTS Line Interface

Standards of the POTS Line Interface

Major standards of the voice line interface are as follows:

- ITU-Q552: It defines transmission specifications of the Z interface.
- ES 201970: It defines basic hardware features of the voice interface.
- YD751 - Network Entry Checking Methods for Telephone Exchange Equipment: It describes the voice interface standards defined by China.

The voice technologies have gone through a long period of development. Almost every country has its own standards. The preceding standards are related to the basic features of service boards. Special features must be tailored to meet requirements of different countries.

Basic Features of the Interface

Basic features of the Z interface supported by the voice interface board of the access device are summarized as follows:

- Battery feeding (B)
  - Batter feeding refers to the supply of the voltage and the current to terminals (such as telephones) to ensure the normal operation of terminals.
  - When the telephone is in the on-hook state, the voltage of the board of the access device is generally 48 V. When the telephone is in the off-hook state, the board of the access device supports the constant-current feeding at 20 mA, 25 mA, or 30 mA. The feeding current can be configured according to the actual requirement.
  - The off-hook feeding of the port can be automatically adjusted. If the length of the loop is short, the port is fed with the constant current. If the length of the loop is long, the port automatically adjusts the loop current based on the preset threshold. This design ensures the compliance with the related standards and optimizes the power consumption of the port.
  - If the feeding current is 25 mA and the voltage is -48 V, the feeding current is equal to or larger than 25 mA when the loop resistance is less than 1200 ohm, and the feeding current is larger than 18 mA when the loop resistance is 1800 ohm.
  - The boards of the access device also support the 40-mA feeding current. The 40-mA feeding current increases the power consumption of ports and thus is not recommended. If the 40-mA feeding current is configured, the number of ports configured with the 40-mA feeding current cannot exceed five on each board.
- Ringing (R)
  - Ringing refers to the supply of the ring current to telephones so that telephones can ring to inform subscribers of incoming calls. The service boards of the device are designed with the balanced ringing feature.

  - The concept of the balanced ringing is put forward based on the concept of the traditional imbalanced ringing. The traditional imbalanced ringing is classified into two types: (1) In a subscriber line, A line is 0 V and B line is -48 V DC overlaid with the 75 Vrms AC signals. (2) In the subscriber line, A line is -48 V and B line carries the 75 Vrms AC signals. In the case of the balanced ringing, both A and B lines of the subscriber line have the AC signals. The AC signals of the A and B lines are of the same frequency and opposite phases, that is, differential signals. The frequency of signals in the case of the balanced ringing can be set to 16 Hz, 25 Hz, or 50 Hz.
  - The amplitude of the ringing current can reach up to 70 Vrms. The amplitude of the ringing current on a terminal can exceed 35 Vrms if the line impedance is 1400 ohm (5-km lines with the core diameter of 0.4 mm) and the terminal impedance is 4000 ohm. The amplitude of the 50-Vrms ringing current is configurable. This configuration is mainly applicable to the short loop with a length less than one kilometer, aiming to substantially reduce the power consumption of ringing on the ports.
  - The DC offset provided by service boards can reach 20 V, which ensures reliable ringing when the distance is long.
  - The break-make ratio of the ringing current can be configured to meet requirements of different carriers in the world.

- Over-voltage protection (O)

  Over-voltage protection is one of the interface protection measures.

- Supervision (S)

  Supervision refers to the detection of telephone state, such as on-hook, off-hook, and off-hook in the ringing state. The terminal state can be learned through detection. The terminal state detection is the basis of some calls.

- Code/Decode (C)

  Coding/Decoding refers to the process that analog signals of the subscriber line are converted into digital signals and compressed according to the A/U law.

- H - Hybrid circuit

  Hybrid circuit refers to the conversion from the 2-wire analog interface to the 4-wire digital interface on the board and implementation of the balanced matching with the impedance of the subscriber line.

- Test (T)

  For details about the test function, see Voice Test and Maintenance.

Interface Impedance, Transmission Specifications, and Gain

The voice interface board of the access device supports the configuration of the interface impedance and gain.

At present, sixteen common interface impedances can be configured, see as the Table 1-14:

Table 1-14 Lists of common interface impedances

ID	Impedance of port and usage description
0	(200+680|| 100nf): bureau machine in China
1	(200+560|| 100nf): user machine in China

ID	Impedance of port and usage description
2	600ohm: a common interface
3	(150+510|| 47nf): interface of Russian
4	(220+820|| 115nf): widely used in countries like Germany
5	(220+820|| 120nf): widely used in Germany
6	900ohm: seldom used
7	(800|| 50nf): interface of Brazil
8	(Zin=87+1052||228nF+229||28.4nF, Zload=93+615|| 471nF+179||495nF+244||32nF): interface of BT0
9	(Zin=370+620|| 310nf,Zload=600): interface of HK_BT3
10	(Zin=270+264|| 357nf+1434|| 265nf,Zload=600): interface of HK_BT5
11	(BT0 without AGC): interface of BT1
12	(Zin=87+1052||228nF+229||28.4nF, Zload=270+264|| 357nF+1434||265nF): Interface of BT2
13	(Zin=87+1052||228nF+229||28.4nF, Zload=164+162|| 363nF+1227||350nF): interface of BT3
14	(Zin= 270+750|| 150nf): a common interface widely used in Europe
15	(Zin= 370+620|| 310nf): interface of New Zealand

The interface transmission gain is also configurable. The send gain is generally in the range of +4 dB and -6 dB and the receive gain is in the range of 0 dB to -12 dB. The gain can be configured at the step of 0.5 dB.

The transmission specifications of the service boards are fully compliant with the ITU-Q522 test requirements. If the interface impedance is not one of the preceding eight types, independent software can be developed to support the interface impedance.

Digit Collection

The voice interface board of the access device supports the pulse-based digit collection.

Old-fashioned telephones generally adopt the pulse dialing mode, while new telephones adopt the DTMF dialing mode. Most telephones support the pulse dialing mode.

The service boards support the pulse-based digit collection at the speed of 8 pps to 12 pps. The break-make ratio is in the range of 50% and 80%. The interval of pulses is configurable and is in the range of 100 ms and 2 s. The default interval of pulses is 300 ms.

The DTMF digit collection is completed by the DSP instead of the service boards.

Charging Signals

Service service boards support three charging modes, namely polarity reverse, 12/16KC, and counter impulse delivery.

- Polarity reverse: The voltage polarity between A and B lines of the subscriber line is reversed. Some terminals detect this type of reverse for charging purpose.
- 12/16KC: The service board sends the 12000 Hz/16000 Hz sine AC signals at a specific interval to the terminals.
- Counter impulse delivery: The service board sends pulse signals to the terminals. Charging is implemented based on the pulse signals.

All ports of the service board support both the fast and slow polarity reverse features. Fast polarity reverse is generally completed within 3 ms, which meets the time requirements of polarity reverse of some telephones. Slow polarity reverse is generally completed within 80 ms. It can substantially reduce the interference to the line during the polarity reverse and is compatible with the DSL transmission on the same line.

The service board supports the 12/16KC charging. In the 12/16KC charging mode, the amplitude of the 12/16KC signals is configurable. The amplitude can be set to 0.45 Vrms, 0.775 Vrms, 1 Vrms, 1.5 Vrms, 2 Vrms, or 2.5 Vrms. The maximum value is 2.5 Vrms (200 ohm). In addition, the break-make ratio of KC signals is also configurable. By default, the Make duration is 100 ms and the Break duration is 300 ms. Both the Make duration and the Break duration range from 10 ms to 500 ms.

The service board also supports the counter impulse delivery charging. Some attributes of this charging mode, such as the pulse width and number of pulses sent per minute, are configurable.

Current Reduction of Locked Ports

When a phone connected to a port is in off-hook state for a long time but the conversation is not going on, the service board can lower the current of the port to less than 12 mA to reduce the power consumption of the port.

Short Loop Feeding

When the length of the line is short, the service board uses the low voltage for feeding to reduce the power of the port. When the length of the line becomes long, the service board automatically uses the voltage higher than the previous low voltage to meet the application requirement.

Power Cut-off

Feeding of ports that are not allocated with numbers can be cut off to reduce the power consumption of the ports.

On-Hook Transmission

Service boards support the on-hook and off-hook transmission functions, such as the caller identification display service and the fixed network short message service.

Ringer Equivalence Number

Ringer equivalence number (REN) refers to the number of telephones that can be connected to the same port.

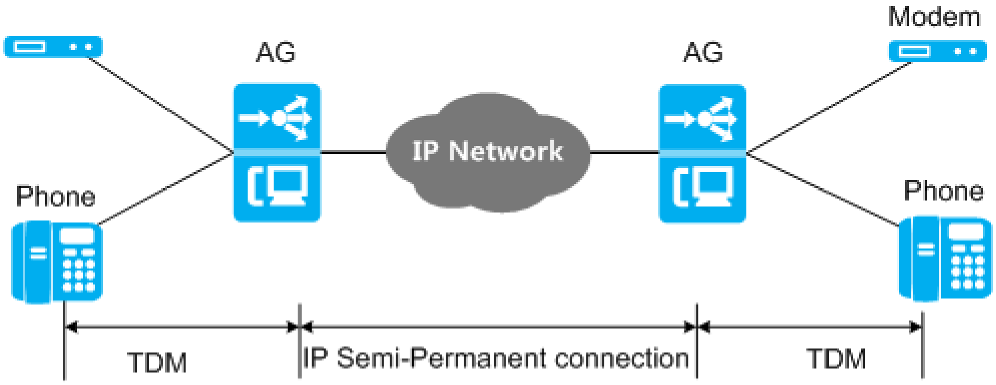
1.9.5 POTS IP SPC

The semi-permanent connection (SPC) exclusively occupies a constant voice channel to meet the communication requirements and ensure the communication quality for special and key access users. To configure an IP SPC, set up an IP direct connection between the two ends of the voice service. In this manner, the voice media data can be directly transmitted to the peer end.

Networking Description

Figure 1-101 shows the networking of a POTS IP SPC. An IP SPC is set up between the POTS port on AG 1 and POTS port on AG 2. Users under these two POTS ports are permanently online and can communicate with each other without dialing a number or any ringing.

Figure 1-101 Networking of a POTS IP SPC



Application Scenarios

A POTS IP SPC is mainly used in the following two scenarios:

- Special modems are connected to a POTS port and these modems require direct communication without dialing and require online permanently.
- Two phones are online permanently without dialing. This scenario may be applied to some special dispatching phones.

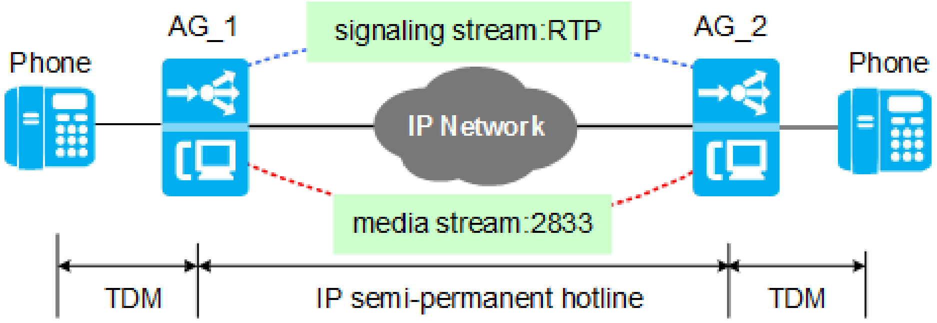
1.9.6 POTS IP SPC Hotline

The semi-permanent connection (SPC) hotline exclusively occupies a constant voice channel to meet the communication requirements and ensure the communication quality for special and key access users. To configure an IP SPC hotline, set up an IP direct connection between the two ends of the voice service. In this manner, the voice media data can be directly transmitted to the peer end. This is mainly used in the situation where two phones are online permanently without dialing. This scenario may be applied to some special dispatching phones.

Networking Description

Figure 1-102 shows the networking of a POTS IP SPC hotline. An IP SPC hotline is set up between the POTS port on AG_1 and POTS port on AG_2. Users under these two POTS ports are permanently online and can communicate with each other without dialing a number. The voice signals on POTS lines are carried through RTP streams when transmitted over the IP network. The call signaling on POTS lines, such as offhook and onhook signaling, is carried through RFC2833 packets.

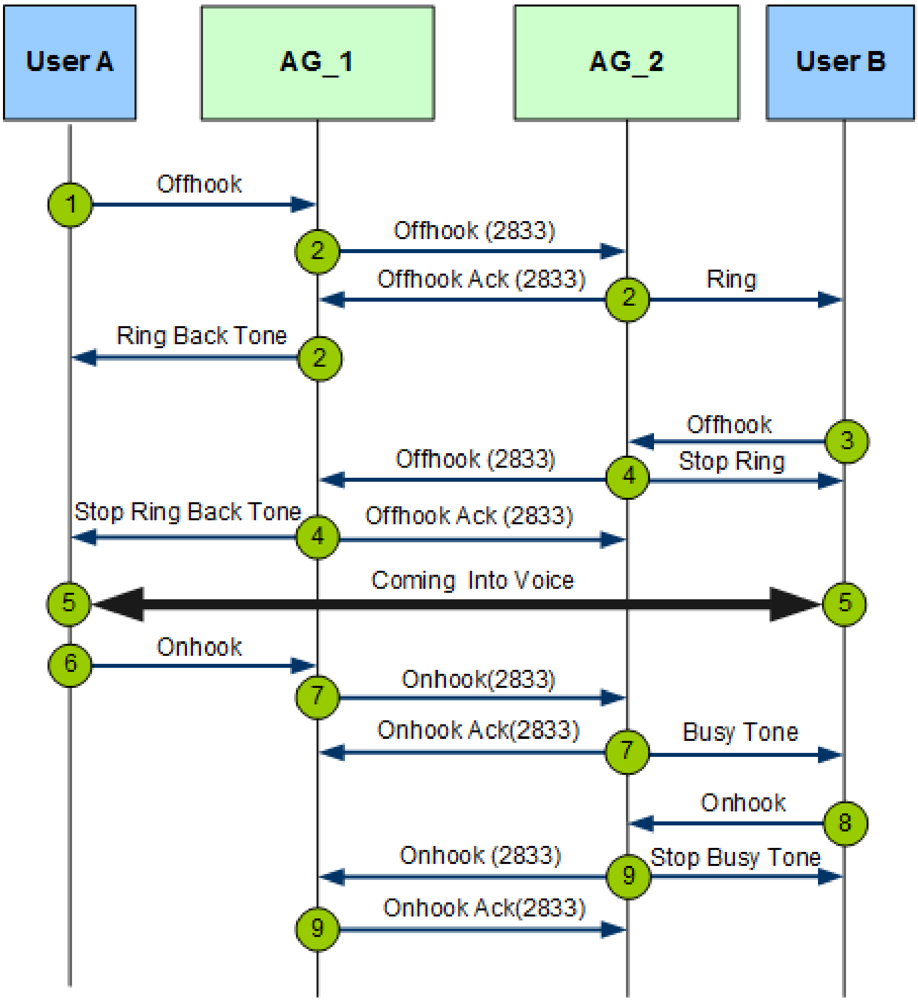
Figure 1-102 Networking of a POTS IP SPC hotline



Call Setup and Release Flow for POTS IP SPC Hotline Service

Figure 1-103 shows the call setup and release flow for POTS IP SPC hotline service.

Figure 1-103 Call setup and release flow for POTS IP SPC hotline service



Call setup and release flow for POTS IP SPC hotline service is as following: (User A is the caller and user B is the callee.)

1. User A picks up the phone.
2. AG_1 detects the offhook signal of user A and sends the offhook signal through an RFC2833 packet to AG_2. AG_2 sends an offhook acknowledgement message to AG_1 through RFC2833, and at the same time plays the ringing tone to user B. After receiving the offhook acknowledgement message, AG_1 transmits the ringback tone to user A.
3. User B picks up the phone.
4. AG_2 detects the offhook signal of user B and sends the offhook signal through an RFC2833 packet to AG_1 and stops the ringing tone to user B. AG_1 sends an offhook acknowledgement message to AG_2 through RFC2833, and at the same time stops the ringback tone to user A.
5. The call is established between user A and user B.
6. User A hangs up the phone.
7. AG_1 detects the onhook signal of user A and sends the onhook signal through an RFC2833 packet to AG_2. AG_2 sends an onhook acknowledgement message to AG_1 through RFC2833, and at the same time plays the busy tone to user B.

8. User B hangs up the phone.
9. AG_2 detects the onhook signal of user B and sends the onhook signal through an RFC2833 packet to AG_1 and then stops the busy tone to user B. AG_1 sends an onhook acknowledgement message to MSAN_2 through RFC2833. The call is released.

1.9.7 POTS Access Specifications

Item	Specifications
CAPS	12

1.10 ISDN Access

The integrated services digital network (ISDN) is a CCITT standard, providing integrated transmission service for voice, video, and data. The ISDN enables the voice, video, and data to be transmitted on the data channel simultaneously.

1.10.1 Introduction to ISDN

Definition

The integrated services digital network (ISDN) is a CCITT standard, providing integrated transmission service for voice, video, and data. The ISDN enables the voice, video, and data to be transmitted on the data channel simultaneously.

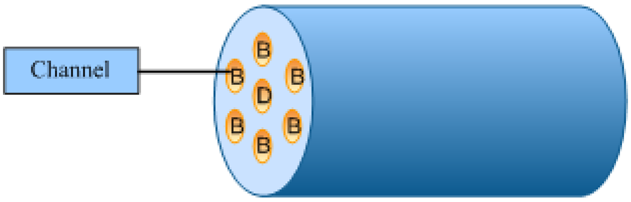
The ISDN supports two types of services:

- Basic rate interface (BRI): provides a rate of 144 kbit/s, including two B channels and one D channel. The rates of B and D channels are 64 kbit/s and 16 kbit/s, respectively.
- Primary rate interface (PRI): provides a rate of 2.048 Mbit/s, including 30 B channels and 1 D channel. The rates of both B and D channels are 64 kbit/s.

ISDN networks support B and D channels, shown in Figure 1-104.

- B channels are used for carrying services.
- D channels are used for transmitting call control signaling as well as maintenance and management signaling.

Figure 1-104 ISDN channels



Purpose

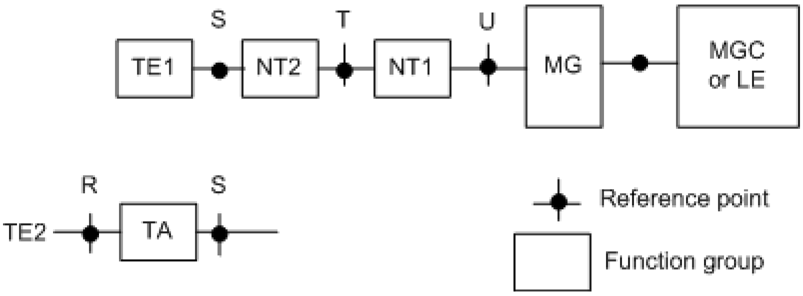
The ISDN access on the media gateway provides integrated transmission services, such as voice, video, and data for the users.

1.10.2 ISDN Protocol Model

ISDN Reference Model

Figure 1-105 shows the ISDN reference model.

Figure 1-105 ISDN reference model



NOTE

- Network Termination 1 (NT1) operates at Layer 1 (physical layer) of the OSI model and implements the physical and electronic specifications into the ISDN network.
- Network Termination 2 (NT2), an intelligent device, functions as a terminal control device, such as a private automatic branch exchange (PABX) or a LAN router, and operates at Layer 2 and 3 of the OSI model.
- Terminal Equipment Type 1 (TE1) is the standard ISDN device having the standard S port, such as an ISDN phone and G4 fax machine. It can be directly connected to NT2 or NT1.
- Terminal Equipment Type 2 (TE2) is a non-standard ISDN device, and it cannot be directly connected to NT2 or NT1 but must be connected to the S port through the TA.
- Terminal Adapter (TA) connects a non-ISDN terminal (TE2) to the user-network interface (UNI) of the ISDN.
- U reference point, also called the U port, is the line interface locates between the ISDN BRA network and user. Digital signals are transmitted through twisted pairs through the coding (such as 2B1Q coding) defined by the U port.
- S reference point, also called the S port, is the line interface locates between the ISDN terminal (TE1 or TA) and NT.
- T reference point locates between NT1 and NT2. If there is no NT2, S referent point and T reference points are combined as S/T reference point, also called the S/T port. It uses 4–wire for transmission, such as a common network cable.
- R reference point locates between the TA and TE2 (non-ISDN standard device) and provides interfaces (the RS-232 interface for PCs and X.25 interface for X.25 devices) that allow the non-ISDN standard device to access the ISDN.

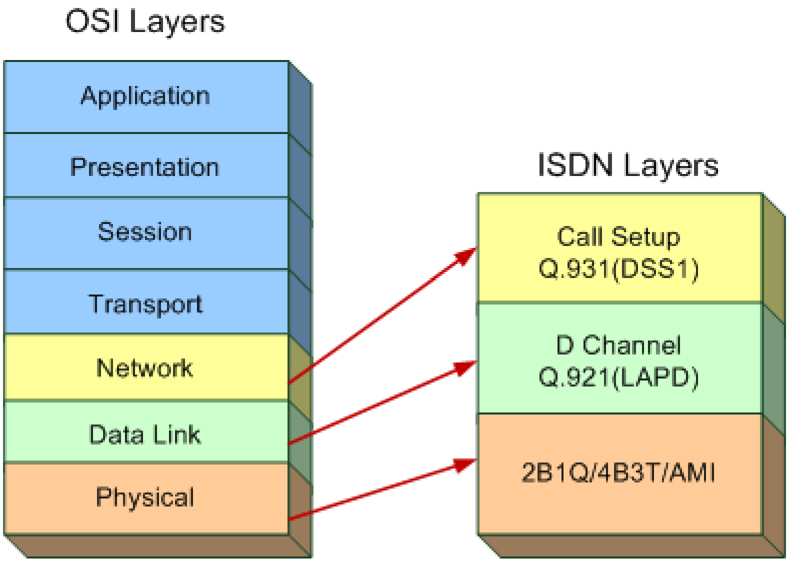
The ISDN user accesses the MA5600T/MA5603T/MA5608T through the U reference point. The actual terminal on the user side may support NT1, NT2, and TE1 functions at the same time. When VoIP is used for upstream transmission, the IUA protocol is used to load the Q.931 call signaling of the ISDN between the MG and MGC, and the H.248 protocol or MGCP signaling is used to control the media connection on the MG.

ISDN Protocol Stack Model

Figure 1-106 shows the mapping relationship between the ISDN protocol and OSI model. Layers in the ISDN protocol stack map the physical layer, data link layer, and network layer in the following OSI model.

- ISDN physical layer: For users, the ISDN physical layer is on S reference point or T reference point. This layer has the following major functions: coding, full-duplex transmission, channel multiplexing, port activation and depolarization, feeding, and termination identification. This layer can multiplex multiple links at the data link layer and use AMI, 4B3T and 2B1Q for coding.
- ISDN data link layer: ISDN does not define Layer 2 protocols dedicated to B channels. Any Layer 2 protocols can be used between two communicating devices after negotiation as long as they can transparently transmit data on B channels. The link access procedure on the D channel (LAPD) protocol defined in Q.921 (a reliable transport protocol) is used for D channels, which is mainly used to carry messages and data generated by Layer 3 entities.
- ISDN network layer: ISDN does not define Layer 3 protocols dedicated to B channels. Layer 3 protocol Q.931 for D channels is mainly used to control and manage connection setup and release on B channels.

Figure 1-106 Mapping relationship between the ISDN protocol stack model and OSI model



ISDN Protocol Processing Model

Figure 1-107 and Figure 1-108 show the ISDN protocol processing model. ISDN involves the following protocols:

- Q.921: Defines LADP. It is a reliable transport protocol.
- Q.931: Defines the procedure of processing and controlling messages and state machines that are used for calls (including circuit switching calls and packet switching calls) between the user-side device and network-side device.
- SCTP: A transport protocol in the SIGTRAN protocol stack, which is a reliable transport protocol on top of protocols (such as IP) providing unreliable transmission services.

SCTP transmits acknowledged, error-free, and repetition-free data and ensures real time transmission to some extent.

- IUA: ISDN Q.921 user adaption layer protocol.

Figure 1-107 H.248 protocol processing model

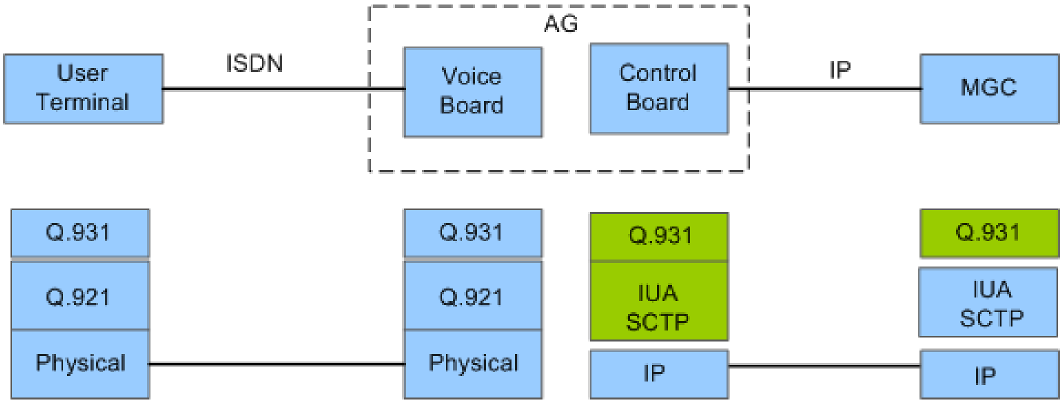
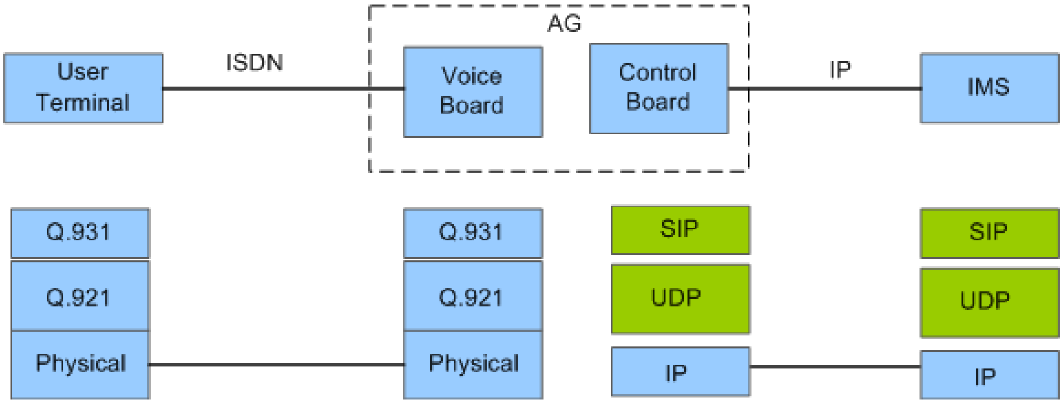


Figure 1-108 SIP protocol processing model



Exchange between protocols

For the H.248 protocol:

- Data transmission from ISDN user terminals to voice boards uses Q.921 and Q.931.
- Voice boards terminate Q.921 messages and send Q.931 messages to the CPU of the control board using the master-slave serial port communications protocol. Then the CPU of the control board uses IUA to packetize Q.931 signaling carried on SCTP links and sends the packed signaling to the MGC through the LAN switch. In this case, Q.921 is not used between the AG and MGC; instead, Q.931 and IUA are used.
- The MGC restores the IUA packets to Q.931 signaling. Also, the MGC sends Q.931 signaling to the peer end through SCTP links. This is the entire process of ISDN call signaling.

For the SIP protocol:

- Data transmission from ISDN user terminals to voice boards uses Q.921 and Q.931.

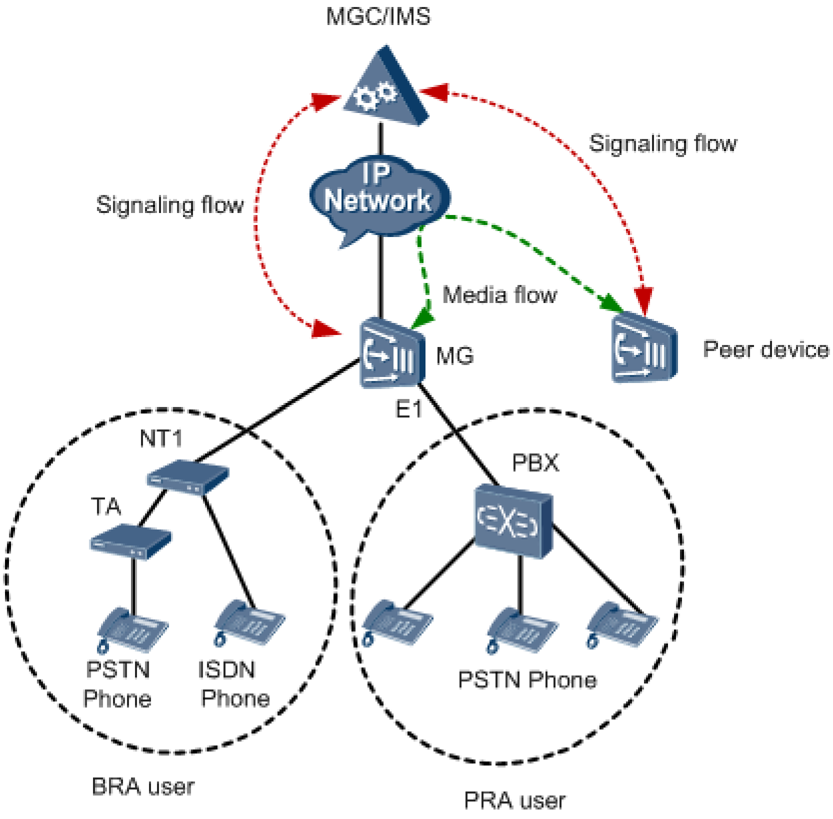
- The AG terminates Q.921 and Q.931 message, converts them into SIP messages and sends SIP messages to the IMS. Then the IMS sends SIP messages to the peer end. This is the entire process of ISDN call signaling.

1.10.3 Call Flow of ISDN

ISDN System Structure

Figure 1-109 shows the ISDN system structure.

Figure 1-109 ISDN System Structure



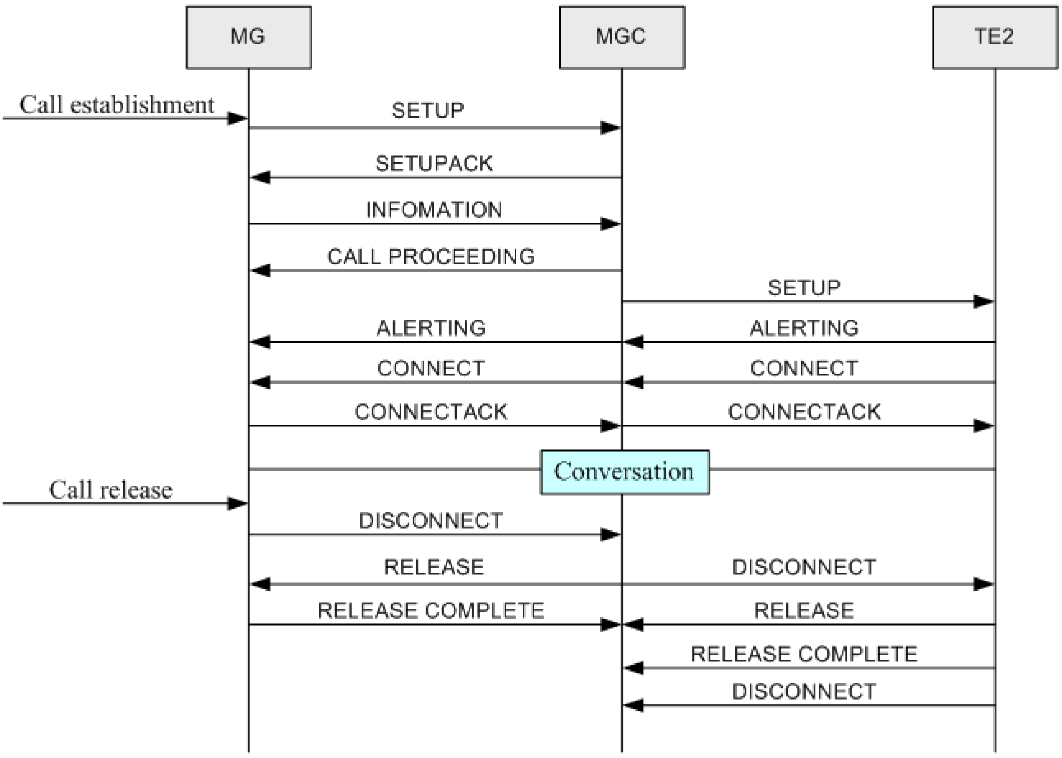
The ISDN users include the BRA users and PRA users.

- The BRA users can connect the ISDN telephone with the NT1 directly, or connect the common telephone through the TA. On the MG side, the BRA users access to the network through the BRA port. Connect the NT1 and MG with the ordinary telephone line.
- The PRA users access the network through the E1 port with the PBX. Connect the PBX and the gateway with the E1 cable.

ISDN Call Control Process

Figure 1-110 shows the ISDN call control process.

Figure 1-110 ISDN call control process



The call process includes two sections: call setup and call disconnection.

- The call setup process is as follows:
  a. The host hooks off and initiates a call setup.
  b. The softswitch responds "SETUP_ACK", and applies more call information, such as the called number.
  c. The calling party dials, and the number is carried by the primitive IMFORMATION to the softswitch.
  d. The softswitch responds "CALL PROCEEDING", and the call is setting up.
  e. The softswitch applies sending setup to the called party to set up a call.
  f. After receiving the call, the called party starts ringing and sends "ALERTING". If the "ALERTING" reaches the calling party, the call is connected.
  g. The called party hooks off and sends "CONNECT". If the "CONNECT" reaches, the call is connected.
  h. The calling party responds "CONNECT_ACK". The call setup is complete.
- The call disconnection process is as follows:
  a. One party hooks on, and sends "DISCONNECT".
  b. The softswitch sends "DISCONNECT" to the other party, and sends "RELEASE" to the party who hooks on.
  c. The party who hooks on finishes the call disconnection, and sends "RELEASE_COMPLETE" to the softswitch.
  d. After receiving the disconnection, the other party sends "RELEASE" to the softswitch.
  e. The softswitch responds "RELEASE_COMPLETE".

f. The other party hooks on, and sends "DISCONNECT". The call disconnection is complete.

1.10.4 The Principles of ISDN BRA

Figure 1-111 and Figure 1-112 show the principles of the ISDN BRA.

Figure 1-111 H.248 protocol principles of the ISDN BRA

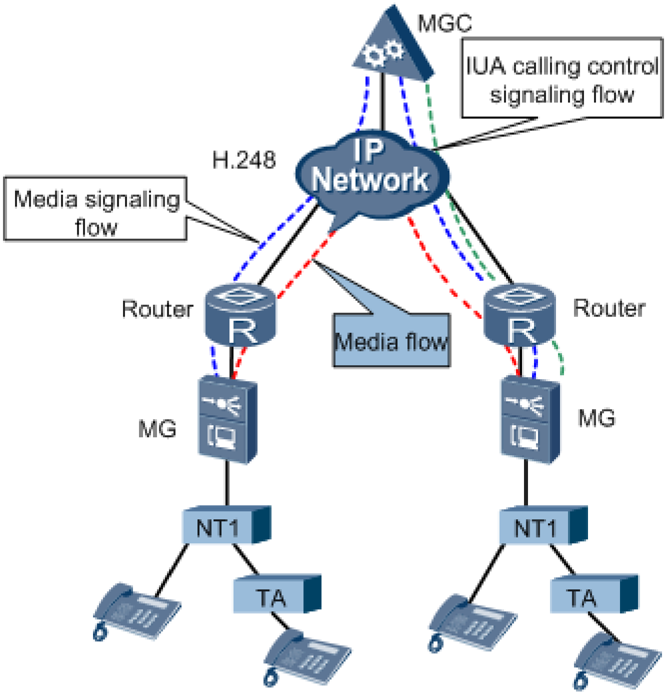
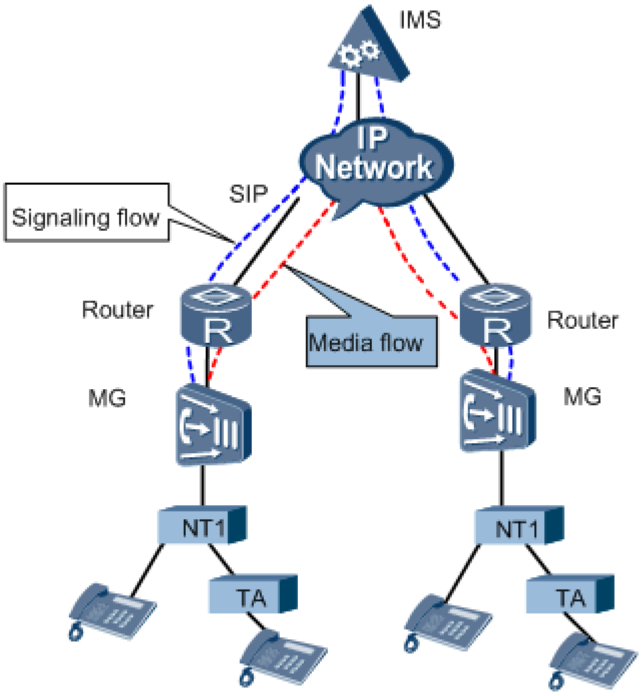


Figure 1-112 SIP protocol principles of the ISDN BRA



User Access

Entering the AN from the MG side, the BRA user call from the deactivated state experiences four stages: activation, TEI application, Layer 2 link setup, and Layer 3 call control. If the port terminal is activated, the TEI is distributed, or the Layer 2 link is set up, skip to next stage.

Call Control

- For the H.248 protocol or the MGCP protocol:
  - According to the signaling round-trip control, the call signaling on the MG is sent to the softswitch through the IUA (as the green line in the figure). The softswitch delivers the media control information through the H.248 protocol/MGCP protocol, and controls the resources on the MG (as the blue line in the figure), such as the B channel, context, and terminal.The gateway does not process the primitive Q.931 but takes out the primitive terminal Q.931 from the Q.921 message, encapsulates the Q.931 to the IUA message, and then sends to the softswitch. Resources are not assigned to the primitive Q.931.

NOTE

The context is supported only when the protocol is H.248.

  - Create an IUA service environment on the MG and MGC sides. Bear the Q.931 signaling on the ISDN BRA service board to the SCTP link, pack the signaling through the IUA protocol stack, and then send the packet to the MGC. Switch the Q.931 signaling on the MGC side. The MGC sends the Q.931 signaling to the peer end through the SCTP link to perform ISDN signaling call.
- For the SIP protocol:
  - The ISDN service is provisioned in the IMS network by mapping and interaction between the SIP signaling and the DSS1 signaling.

   - The gateway converts the ISDN message to the SIP message and then the ISDN uses the SIP protocol for call control. The gateway between the ISDN terminal and the IMS implements signaling conversion and service control. Specifically, the gateway translates the DSS1 signaling to the SIP signaling and then sends the signaling to the IMS. Generally, the IMS transparently transmits the SIP signaling to the peer ISDN user. Then, the peer ISDN user translates the SIP signaling to the DSS1 signaling and then sends the signaling to the ISDN terminal. This process enables two ISDN terminals to be connected with each other and therefore the ISDN service is provisioned.

Working Mode

The BRA working modes include point to multipoint (P2MP) and point to point (P2P).

- In the P2MP mode, one NT1 can connect to multiple terminals. Multiple Layer 2 links can be created at the same time, and up to two users can call simultaneously. If no call service exists, the system can be deactivated automatically to save the power and it also supports longtime activation.
- In the P2P mode, one NT1 can connect to one terminal only. The Layer 2 link is always set up to ensure the service bearing at any moment. No matter whether the call service exists, the link is activated.

Terminal Power Supply Mode

The BRA power supply is to provide power for the terminal. Two terminal power supply modes are provided:

- Local power supply: The terminal applies battery or connects to the power supply.
- NT1 power supply: The terminal uses the NT1 power supply only. The NT1 power supply falls into two categories:
  - Local power supply: The NT1 connects to the local power supply.
  - Gateway power supply: Configure the remote power supply attribute of the BRA port on the gateway.

Terminal Identifier Distribution

In the P2MP mode, if the physical line of the BRA user is activated, one BRA port can connect multiple terminals. A terminal equipment identifier (TEI) is needed to identify the terminal.

The TEI can be specified by the terminal, or distributed on the network side.

- The TEI that the terminal specifies ranges 0-63.
- The TEI on the network side is distributed by the subscriber board, ranging 64-126.
- The 127, as a multicast TEI, is used when the BRA user is called (all the users under the same port share the same telephone number). When the destination terminal is unknown, the connections to all the terminals are initiated.
- In the P2P mode, the terminal TEI is 0.

1.10.5 The Principles of ISDN PRA

The PRA call process is the same as the BRA call process. For the BRA call process, refer to 1.10.4 The Principles of ISDN BRA.

- One PRA user has 32 time slots with the rate of 64 kbit/s. Among the 32 time slots, time slots 1-15, 17-31 are for the B channel, time slot 16 is for the D channel, and time slot 0 is for frame synchronization.
- For a PRA user, the TEI of the L2 link is 0.
- For a PRA user, the working mode and power supply mode are not included. The terminal is powered by the PBX.

1.10.6 ISDN Standards and Protocols Compliance

This topic provides the reference documents of the ISDN:

- ITU-T Q.920 ISDN user-network interface data link layer General aspects
- ITU-T Q.921 ISDN user-network interface - Data link layer specification
- ITU-T Q.930 Digital Subscriber Signalling System No.1 (DSS 1) -ISDN User-Network Interface Layer 3 - General Aspects
- ITU-T Q.931 ISDN user-network interface layer 3specification for basic call control
- ITU-T H.248 Media gateway overload control package
- RFC3435 Media Gateway Control Protocol (MGCP) Version 1_0
- RFC3660 Basic Media Gateway Control Protocol (MGCP) Packages
- RFC3661 Media Gateway Control Protocol (MGCP) Return Code Usage
- ITU-T G.961 Digital transmission system on metallic local lines for ISDN basic rate access

1.11 R2 Access

R2 access enables the MA5600T/MA5603T/MA5608T to be interconnected with a private branch exchange (PBX) through the R2 signaling and helps to provide access services for users over the G.SHDSL ports and E1 ports. As a type of channel associated signaling (CAS), R2 signaling is the international standard signaling based on E1 digital networks.

1.11.1 Introduction to the R2 Feature

Definition

R2 signaling is a type of channel associated signaling (CAS) and it is also the international standard signaling based on E1 digital networks. Timeslot 16 in the R2 signaling is reserved for transmitting the signaling of the voice channel.

Actually, there is no agreed standard for R2 signaling. ITU-T Recommendation Q.400-Q.490 define the R2 signaling standards but different countries and regions have developed their own standards.

Purpose

The MA5600T/MA5603T/MA5608T connects an R2 PBX to the next generation network, achieving transformation of the public switched telephone network (PSTN) to the NGN.

1.11.2 R2 Principles

R2 signaling is inter-office channel associated signaling, and applies to international/national networks. R2 signaling is specified on both analog and digital transmission systems. R2 signaling contains line signaling and register signaling. Line signaling is available in three forms: DC line signaling, inband single-frequency pulse line signaling, and digital line signaling. For multi-end route transmission, the link-by-link forwarding mode is used. Register signaling can be transmitted in multi-frequency compelled (MFC) mode and dual tone multiple frequency (DTMF) mode.

Line Signaling

Line signaling is primarily used for monitoring the occupation, release and congestion states of a trunk line. Line signaling is classified into analog line signaling and digital line signaling. The MA5600T/MA5603T/MA5608T supports only the digital line signaling which will be described in detail in the following paragraph.

The digital line signaling uses timeslot 16 of the PCM for transmitting line signaling at a rate of 2048 kbit/s. To transmit line signaling of 30 voice channels, 16 frames form a multiframe. Timeslot 16 of frame 0 in the multiframe is used for multiframe synchronization. The first four bits of timeslot 16 in frame 1 correspond to the first voice channel, while the last four bits correspond to the 16th voice channel and so on.

Register Signaling

Register signaling is the signaling transmitted over a voice channel after the line signaling occupies the voice channel. Register signaling, including the selection signaling and service signaling, is used to transmit the control signals for a voice channel connection, such as managing the telephone network and selecting the route and the called party. The MFC register signaling will be described in detail.

The MFC register signaling includes the forward signaling and backward signaling, both of which are consecutive. The forward signaling is used to transmit the address information and control the indication information, while the backward signaling is used for acknowledgement and control. When transmitting a number, the transmit end stops transmitting the forward signaling only after receiving the acknowledgement from the backward signaling. Similarly, the receive end stops transmitting the backward signaling only after confirming that the transmission of the forwarding signaling is stopped. A control period can be divided into four steps, which is listed as follows:

- Step 1: The user side sends the forward signaling.
- Step 2: The network side receives the forward signaling and returns the backward signaling.
- Step 3: The user side receives the backward signaling and stops sending forward signaling.
- Step 4: The network side finds that the transmission of forward signaling is stopped, and stops sending the backward signaling.

The MFC register signaling uses the arithmetic frequency with 120 Hz as the common difference. This section takes the definition in Q.441 as an example, the forward signaling uses the high-frequency group (1380 Hz to 1980 Hz) and selects two out of six frequencies (1380 Hz, 1500 Hz, 1620 Hz, 1740 Hz, 1860 Hz, and 1980 Hz) for encoding. Up to 15 signaling combinations can be formed. The backward signaling uses the low-frequency group (780 Hz to 1140 Hz) and selects two out of four frequencies (780 Hz, 900 Hz, 1020 Hz, and 1140 Hz) for encoding. Up to six signaling combinations can be formed. To expand the signaling capability, the forward signaling is divided into group I forward signaling and group II forward signaling, while the backward signaling is divided into group A backward signaling and group B backward signaling.

- Register signaling must always start with group I forward signaling. Group I forward signaling includes the information of the country code, echo suppressor indicator (I-11) and address signal (number: 1-9).
- Group II forward signaling is the calling party's category signaling sent by an outgoing R2 register. It is used to reply to backward signal A-3 (the address-complete signal) or A-5 (the request signal for a calling party's category), and send the national or international calling information.
- Group A backward signaling is used to acknowledge the group I forward signaling and under some conditions, group II forward signaling, such as acknowledging the calling party's type and group II forward signals.
- Any group B backward signaling acknowledges the group II forward signaling and is always preceded by the address-complete signal A-3. Signal A-3 indicates that the incoming R2 register has received all the required forward signals from the outgoing R2 register.

Register signaling is the in-band signaling (the frequency is within the voice frequency band). Therefore, the register signaling is transmitted over the voice channel.

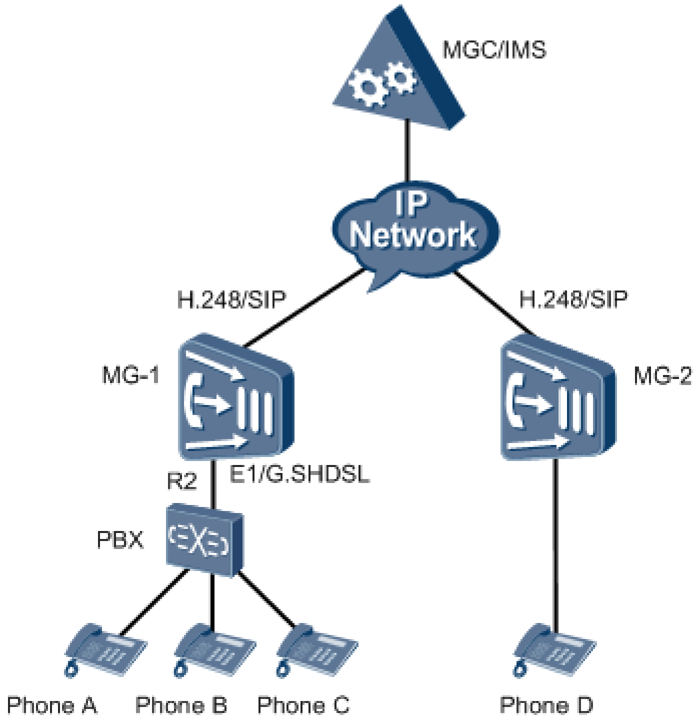
R2 Application in Access Networks

A PBX is connected to the next generation network (NGN) through R2 signaling. Interfaces between an MG and an MGC/IMS use the H.248/SIP protocol, and interfaces between an MG and a PBX use the R2 protocol. Figure 1-113 shows the networking of connecting the R2 PBX to the NGN network.

To connect a PBX to the NGN network through R2 signaling, MGs need to perform the conversion between R2 signaling and H.248/SIP signaling.

- In the upstream direction, the MA5600T/MA5603T/MA5608T terminates R2 signaling transmitted from the PBX, converts R2 signaling to H.248/SIP signaling, and sends the converted signaling to the softswitch.
- In the downstream direction, the MA5600T/MA5603T/MA5608T terminates H.248/SIP signaling transmitted from the softswitch, converts H.248/SIP signaling to R2 signaling, and sends the converted signaling to the PBX.

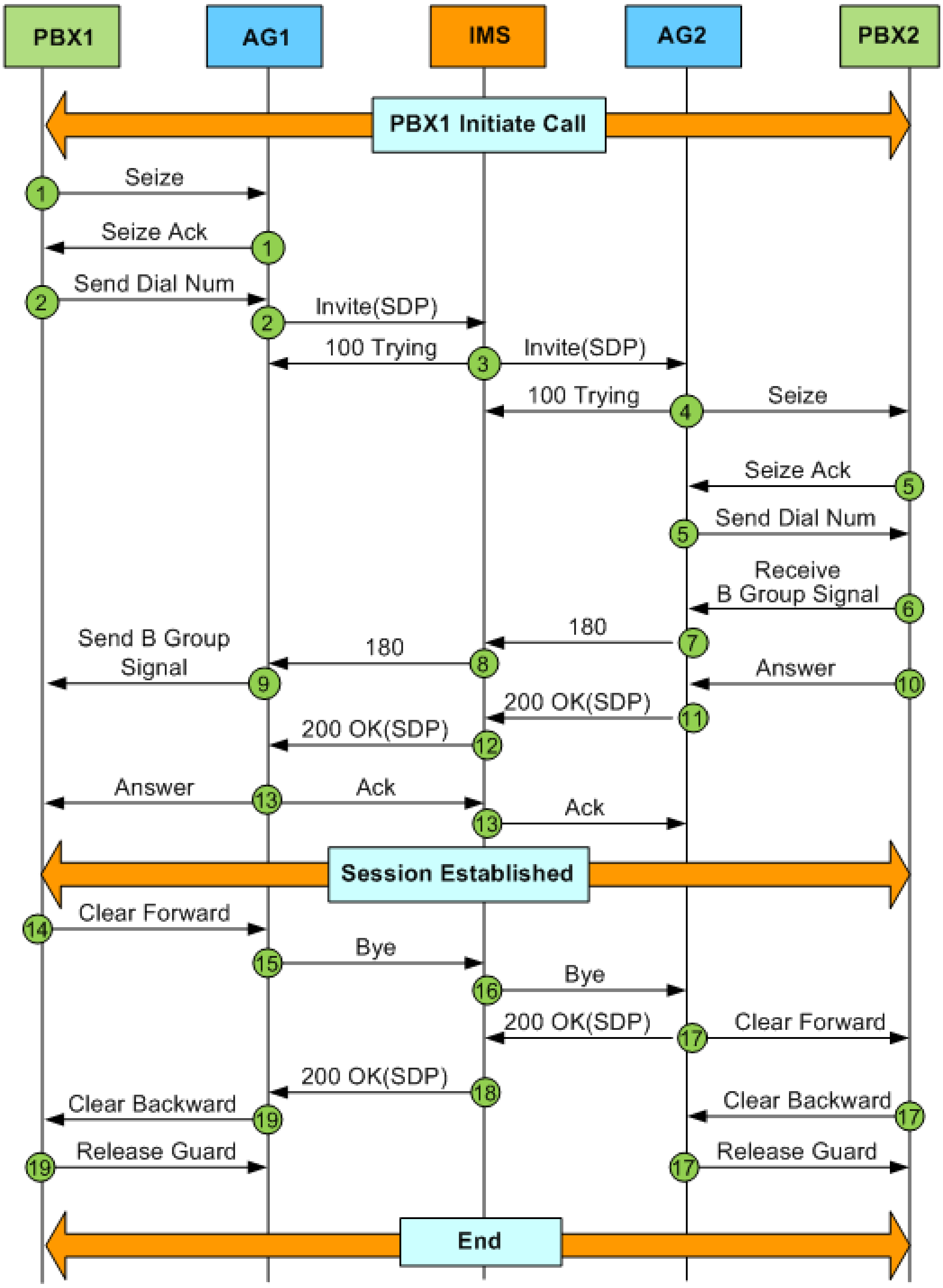
Figure 1-113 Network example of connecting the R2 PBX to the NGN network



1.11.3 Call Flow of SIP R2

R2 call flow manages the call connection between the AG and R2 terminals, and between the AG and IMS, which implements R2 terminal connection to the IMS network. Figure 1-114 shows the call flow.

Figure 1-114 Call flow of SIP R2



Basic R2 call flow:

1. PBX1 sends the Seize event to AG1, and AG1 responds with Seize Ack and starts R2 digit collecting.
2. AG1 sends the Invite message after collecting all digits and matches the digitmap.
3. After receiving the Invite message, the IMS responds with 100 Trying and forwards Invite to AG2.

4. After receiving Invite, AG2 responds with 100 Trying, finds out PBX2 according to the called number, and sends Seize to PBX2.
5. PBX2 responds with Seize Ack and AG2 sends the converted called number to PBX2.
6. PBX2 finds out the callee according to the received called number and responds to AG2 with B signaling and marks callee.
7. AG2 sends response 180.
8. The IMS forwards response 180 to AG1.
9. AG1 responds with B signaling to PBX1.
10. The user under PBX2 picks up the phone and PBX2 sends message Answer to AG2.
11. AG2 sends response 200 to the IMS.
12. The IMS forwards response 200 to AG1.
13. AG1 sends message Answer to PBX1 and responds with Ack to the IMS for communication.
14. The user under PBX1 hangs up the phone and PBX1 forwards message Clear Forward to AG1.
15. AG1 sends message Bye to the IMS.
16. The IMS forwards message Bye to AG2.
17. AG2 responds with message 200 to the IMS and sends message Clear Forward to PBX2. Then, PBX2 responds with Clear Backward to AG2 and AG2 sends Release Guard to PBX2. The outgoing call is terminated.
18. The IMS forwards response 200 to AG1.
19. AG1 sends Clear Backward to PBX1. After AG1 receives Release Guard sent by PBX1, the incoming call is terminated.

NOTE

- The device supports the R2 service connected by E1 lines and G.SHDSL access.
- The R2 service supports only basic calls.
- Dialing modes MFC and DTMF are supported.

1.11.4 R2 Standards and Protocols Compliance

The reference standards and protocols of this feature are as follows:

- draft-manyfolks-megaco-caspackage-01.txt
- draft-laha-megaco-cas-mntc-00.txt
- draft-ietf-megaco-r2package-03.txt
- ITU-T H.248.25 Gateway control protocol: Basic CAS packages
- Specifications of Signaling System R2, Q.400 to Q.490, Blue Book, CCITT

1.12 FoIP

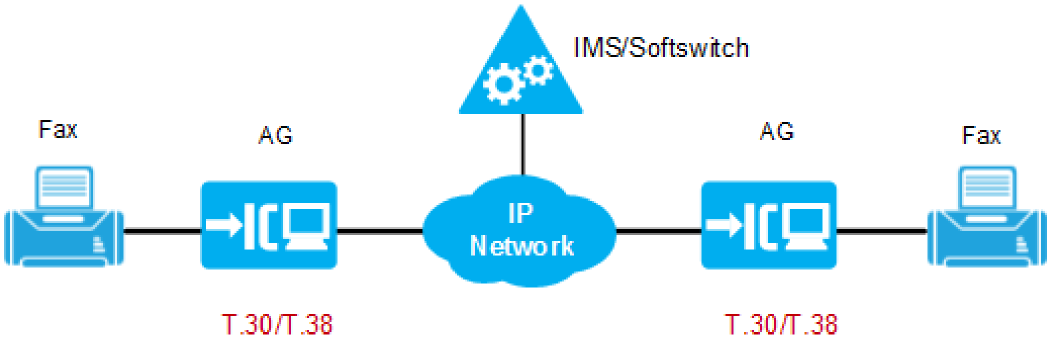
Fax over Internet Protocol (FoIP) is a fax service provided on an IP network or between an IP network and a traditional PSTN. Fax service is a data service that is widely applied on the PSTN network.

1.12.1 What Is FoIP

Definition

Fax over Internet Protocol (FoIP) is a fax service provided on an IP network or between an IP network and a traditional PSTN. Figure 1-115 shows the network application of FoIP.

Figure 1-115 FoIP network application



Basic Process

The basic process of fax can be described as follows:

1. The fax transmitting machine scans a page to obtain image information.
2. The fax transmitting machine digitalizes and compresses the image signals.
3. The fax transmitting machine modulates the image signals into analog signals, and transmits the signals to the fax receiving machine through common subscriber lines (as defined in the T.30 protocol).

Commonly Used Protocols for Fax

The following protocols are commonly used for fax:

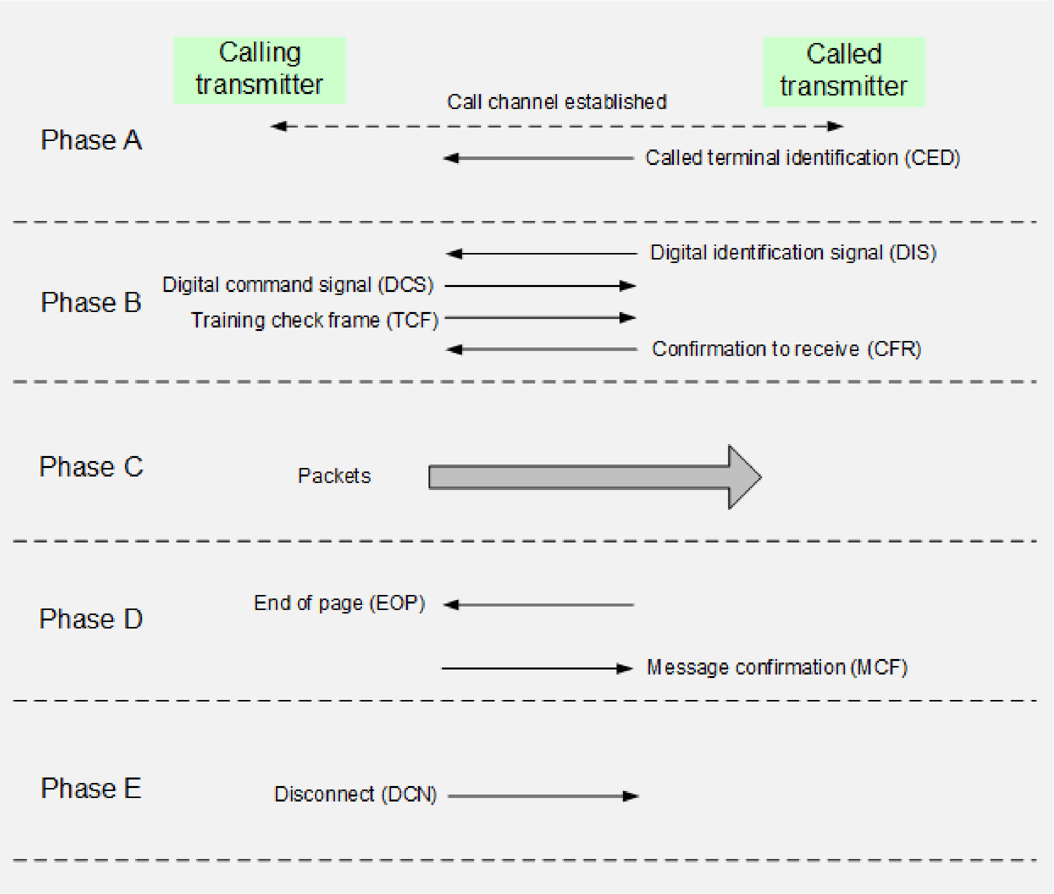
- T.30: It is a fax protocol based on the PSTN network. T.30 defines in detail the process for transmitting fax signals on a PSTN network. It also defines the modulation mode (V.17/V.21/V.27/V.29/V.34) and transmission format (HDLC) of data, and the physical standard for fax signals. The T.30 fax messages and data can be transmitted transparently between AGs. This is called the T.30 transparent transmission mode. The quality of fax in this mode may not be high due to packet loss, delay, and packet disorder on the IP network.
- T.38: It is a protocol that defines the process for carrying fax services over a packet IP network. Serving as a supplementary protocol to T.30, T.38 implements packet encapsulation based on T.30 in order to adapt to IP applications. In T.38 fax service, the T.38 redundancy mechanism is used to ensure service quality.

Typical Fax Process Defined in T.30

T.30 defines 5 phases for a typical fax process, as shown in Figure 1-116.

- Phase A: A call for fax is established. This phase is similar to a telephone call establishment phase.
- Phase B: The earlier phase of the packet transmission process. In this phase, the devices at both ends perform capability negotiation and training.
- Phase C: This is the packet transmission phase and also the packet transmission control phase.
- Phase D: The later phase of the packet transmission process. In this phase, packets are verified, errors are corrected, and multiple pages are continuously transmitted.
- Phase E: The call for fax is released.

Figure 1-116 Typical fax process defined in T.30


Calling transmitter
Called transmitter
Phase A
Call channel established
Called terminal identification (CED)
Phase B
Digital identification signal (DIS)
Digital command signal (DCS)
Training check frame (TCF)
Confirmation to receive (CFR)
Phase C
Packets
Phase D
End of page (EOP)
Message confirmation (MCF)
Phase E
Disconnect (DCN)

1.12.2 Classification of FoIP

Fax over IP (FoIP) is a faxing service provided over an IP network or between an IP network and a traditional PSTN network. The fax machine can be regarded as a special Modem. In the FoIP negotiation, the Modem negotiation is performed before the fax negotiation. FoIP services can be classified based on the transmission real-time performance, or based on the transmission mode.

Classification Based on Transmission Real-time Performance

Based on the real-time performance, FoIP can be classified in store-and-forward FoIP and real-time FoIP. The difference between the two modes lies in whether communication between the gateway and the fax machine is real-time on the IP side. On the PSTN side, communication of the two FoIP modes is real-time.

- Store-and-forward FoIP: In this mode, fax information is stored and then forwarded to the IP network, as defined in the T.38 protocol.
- Real-time FoIP: In this mode, communication during the entire fax process is carried out in real time, as defined in the T.38 protocol.

NOTE

Huawei access gateway (AG) supports real-time FoIP.

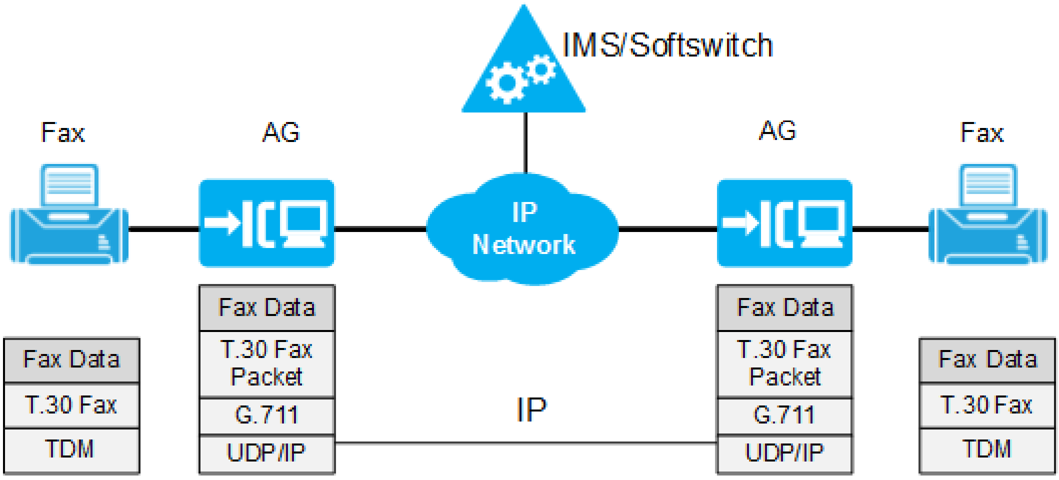
Classification Based on Transmission Mode

According to the transmission protocol used, there are two modes of fax services carried over the IP network: the T.30 transparent transmission mode and the T.38 transmission mode.

T.30 transparent transmission mode

In this mode, T.30-defined fax messages and data are transparently transmitted in an AG or between AGs. Figure 1-117 shows the T.30 transparent transmission mode.

Figure 1-117 T.30 transparent transmission mode



The advantages and disadvantages of the T.30 transparent transmission mode are as follows:

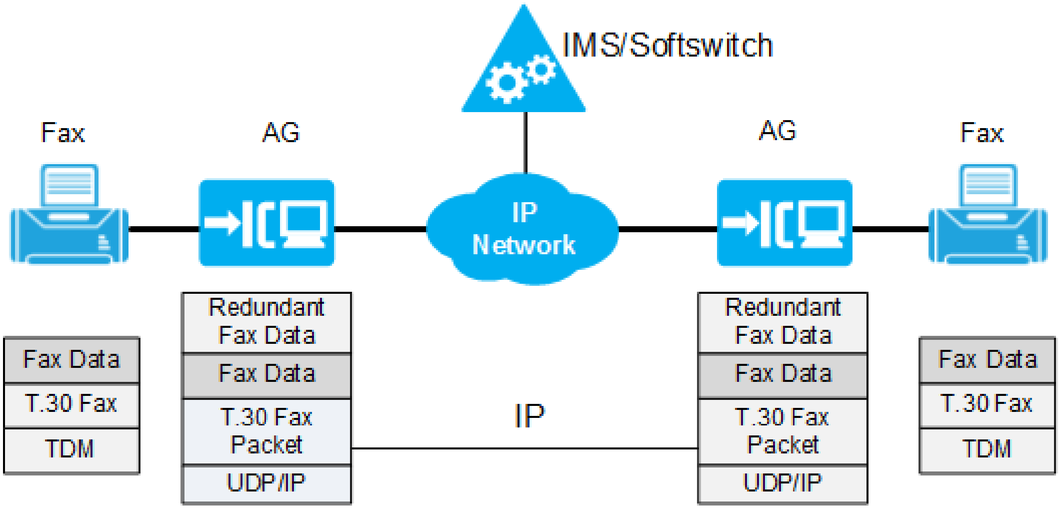
- Advantages: consumes less DSP resources, and is less dependent on the softswitch/IMS.
- Disadvantages: weak resistance against interference from the network, and does not provide reliable guarantee for fax quality. Stability of the T.30 transparent transmission mode can be improved using the RFC 2198 and 10-ms packetization technologies. For more details, see Fax/Modem Quality Enhancement.

T.38 transmission mode

The T.38 transmission mode is shown in Figure 1-118. T.38 fax supports two rate negotiation modes: end-to-end negotiation and local negotiation. The difference between the two negotiation modes lies in whether the rate training signals need to be transmitted from the transmitting AG to the receiving AG.

- When the rate training signals need to be transmitted from the transmitting AG to the receiving AG, it is end-to-end negotiation.
- When the rate training signals are terminated and generated by the transmitting AG, it is local negotiation. If the local negotiation mode is used, the maximum rate supported by the AG should be considered. That maximum rate is reflected by the maximum fax rate supported by the digital signal processor (DSP).

Figure 1-118 T.38 transmission mode



The advantages and disadvantages of the T.38 transmission mode are as follows:

- Advantages: provides a redundancy mechanism for transmitting data packets, and does not have strict requirements on network quality (able to process the fax service even with a 20% packet loss rate on the network).
- Disadvantages: The DSP chip of the AG needs to participate in parsing the T.38 signals. Because there are various types of terminals on the network, interoperability problems may occur.

Differences Between High-speed Fax and Low-speed Fax

The main differences between high-speed fax and low-speed fax include the following:

- Standards applied. High-speed fax applies the V.8 data transmission process, while low-speed fax applies a fax process defined by the T.30 protocol. In addition, some low-speed fax terminals may use earlier standards.
- Range of rates supported. High-speed fax supports a rate range of 2400 bit/s-33600 bit/s, while low-speed fax supports a rate range of 2400 bit/s-14400 bit/s.
- Upstream transmission modes used. High-speed fax can use only the T.30 transparent transmission mode. In other words, for an AG, only the high-speed Modem T.30 transparent transmission mode can be used for fax services. Low-speed fax can use the T.30 transparent transmission mode or T.38 transmission mode, according to data configuration of the fax terminal.

- Error correction requirements. For high-speed fax, the error correction function is a mandatory requirement; for low-speed fax, the error correction function is optional.
- Echo cancellation (EC) requirements. High-speed fax requires the EC function to be disabled because a high-speed fax terminal already has an inherent EC mechanism; low-speed fax requires the EC function to be enabled.

1.13 MoIP

Modem over Internet Protocol (MoIP) is a technology for providing modem services over an IP network or between an IP network and a PSTN network.

1.13.1 What Is MoIP

Definition

The term "modem" is abbreviated from modulator and demodulator. Modem service is a data service that is widely applied on the PSTN network. In its earlier application, modem service is mainly used for point-to-point dialing and Internet dialup. Later, the service scope is expanded to cover point of service (POS) machine connection, alarming, and lottery machine connection.

Modem over Internet Protocol (MoIP) is a technology for providing modem services over an IP network or between an IP network and a PSTN network.

The dialup mode of modems can be pulse dialup and tone dialup.

- In the traditional modem dialup scenario, PSTN users directly dial numbers, and the media stream passes through only the narrowband channel.
- MoIP differs from the traditional modem dialup service in that MoIP users are next generation network (NGN) access gateway (AG) users, and that modem negotiation and media stream transmission between the AG and the trunk gateway are performed based on IP.

MoIP Services

Modem services are usually transmitted using the V.32/V.34/V.90/V.92 protocol. Different modem protocols support different ranges of rates. In practical service application, the transmission rate is usually the result negotiated by the two modems at both ends.

- V.32 supports a maximum rate of 14.4 kbit/s.
- V.34 supports a maximum rate of 33.6 kbit/s. V.34 is the modem protocol most commonly used. It defines a negotiation process that is the same as the high-speed fax process. One application example of V.34 is the POS machine service.
- V.90/V.92 supports a maximum rate of 56 kbit/s and is usually used for Internet dialup services.

Modem services are classified into high-speed modem services and low-speed modem services. The negotiation process of high-speed modem services is the same as that of high-speed fax services.

- Based on the service rate, services with rates lower than or equal to 14.4 kbit/s are low-speed modem services, and services with rates higher than 14.4 kbit/s are high-speed modem services.

- Based on the modem's requirements on the network, modems that require the network device involved to disable the echo cancelation function are high-speed modems, and modems that do not require so are low-speed modems.

1.13.2 Principle of MoIP

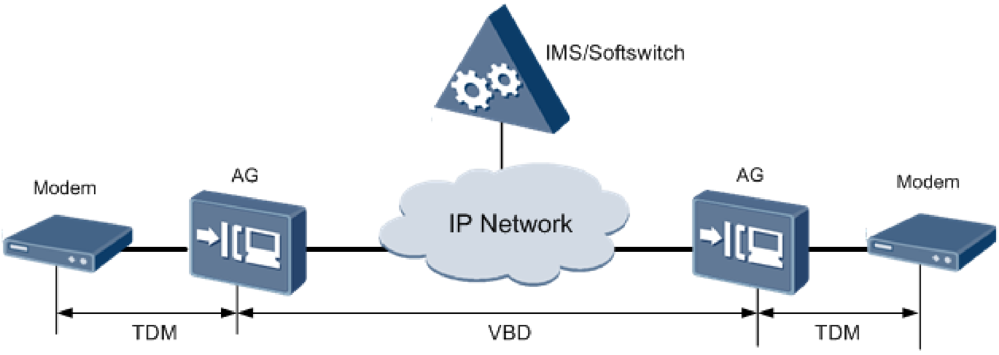
Just like VoIP, MoIP supports the AG-based PSTN-IP-PSTN network structure, and also supports the PSTN-IP network structure.

MoIP Transmission Modes

MoIP has two transmission modes:

- Transparent transmission mode, which is also called the voice-band data (VBD) transparent transmission mode. In this mode, the AG uses the G.711 mode to encode and decode the modem signals, and processes the signals like processing common Real-time Transfer Protocol (RTP) data. In other words, the AG does not process the modem modulation signals, which are transparently transmitted on the IP network through VoIP channels. Figure 1-119 shows the MoIP transparent transmission mode.

  Figure 1-119 MoIP transparent transmission mode

  - High-speed modem: with voice activity detector (VAD) and echo canceller (EC) disabled.
  - Low-speed modem: with VAD disabled and EC enabled.
- Redundancy mode, which is also called the relay mode.

NOTE

Currently, Huawei AGs support only the transparent transmission mode for MoIP.

Enhanced MoIP

The MoIP service carried in transparent transmission mode has high requirements on the bearer network. Jitter, delay, and packet loss on the network will affect modem services significantly. In case of poor network quality, the MoIP service can be enhanced using the RFC 2198 and 10-ms packetization technologies. Using these technologies, the connection success ratio can be improved for modems, and the modem disconnection ratio will be reduced as well. For more details, see Fax/Modem Quality Enhancement.

1.14 IP Z Interface Extension

IP Z interface extension is that the analog interface between an accsee device and a PBX extends to the remote place through the IP network.

1.14.1 Introduction to IP Z Interface Extension

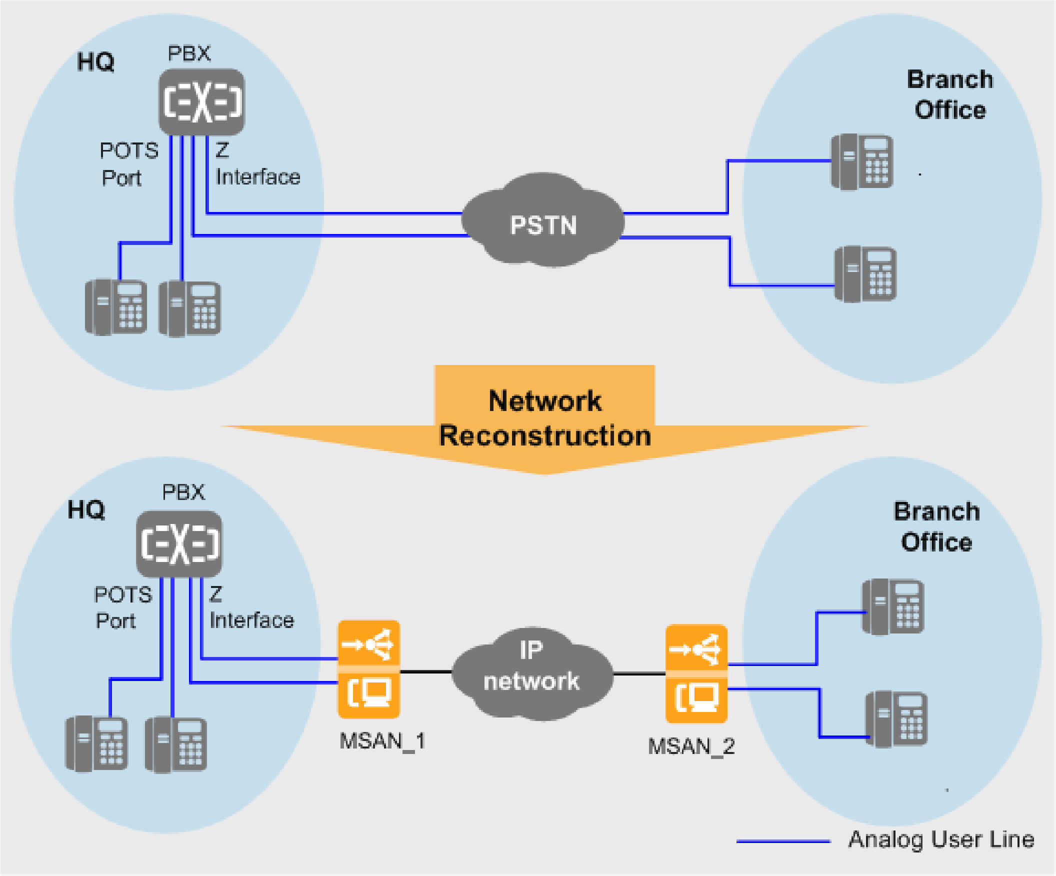
Context

The PSTN network can be used to provision the Z interface extension private line service for the headquarters (HQ) and branch offices of businesses. The service enables users at the branch offices to have the same calling experience as users at the HQ through phones. As devices on existing PSTN networks are approaching the end of their life cycles, many carriers are reconstructing the PSTN networks.

Figure 1-120 illustrates the Z interface extension service network before and after the PSTN network reconstruction. After the reconstruction, the Z interface extension service that used to be carried by the PSTN network is carried by an IP network.

The access devices, one located at the HQ and the other at the branch office, are interconnected through the IP network. The voice signals on POTS lines are carried through RTP packets when transmitted over the IP network. The call signaling on POTS lines, such as offhook and onhook signaling, is carried through RFC2833 packets.

Figure 1-120 Z interface extension service network before and after PSTN reconstruction



Definition

The Z interface refers to the analog interface between an access device and an exchange.

Z interface extension means the extension of POTS user signals. Analog POTS user signals are converted into digital signals at the Tx end, transmitted over the network, and restored into analog signals at the Rx end. In this feature, the extension of the Z interface is carried over an IP network, so the feature is called IP Z interface extension.

Purpose

IP Z interface extension is intended for the following purposes:

- Connect the private line users at the branch offices of an enterprise that has a small volume of analog phone service requirements to the enterprise's local exchange.
- Extend the calls of analog phones under a PBX to a remote location, thereby saving long-distance call fee.
- Connect users at a remote location to a local exchange.

Hardware support

Boards on the HQ-side MSAN (such as MSAN_1 in Figure 1-120) that provide the IP Z interface extension service are the FXO board. Boards on the branch office-side MSAN (such as MSAN_2 in Figure 1-120) for the same purpose are the FXS board.

Limitations

IP Z interface extension is a technology proprietarily owned by Huawei. Therefore, the MSANs on the FXO side and the FXS side must be Huawei MSANs.

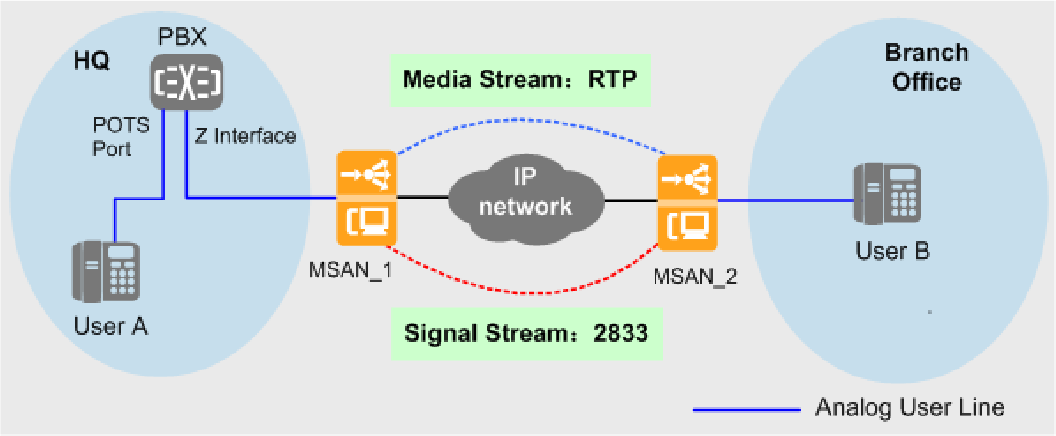
1.14.2 Principle of IP Z Interface Extension

The typical IP Z interface extension service includes a Z interface, the IP transmission network, and a POTS port.

Figure 1-121 demonstrates the implementation principle of IP Z interface extension. In this network diagram, two access devices, MSANs, one located at the headquarters (HQ) and the other at the branch office, are interconnected through the IP network. Such a networking model eliminates the limitations on the line type and line transmission distance, and covers a wider range of users. MSAN_1 at the HQ converts the analog signals of the local exchange (PBX) into digital signals using the analog-to-digital conversion mechanism. The voice signals on POTS lines are carried through RTP packets when transmitted over the IP network. The call signaling on POTS lines, such as signaling for offhook, onhook, hookflash pressing, and ringing, is carried through RFC2833 packets. In this way, the voice signals and call signaling are transmitted using the IP network to the POTS port of the remote access device MSAN_2. MSAN_2 restores the analog signals from the digital signals using the digital-to-analog mechanism, thus extending POTS signals from the Z interface of MSAN_1 to the POTS port of MSAN_2.

The IP Z interface extension services enables user A and user B to have the same user experience when making outgoing and incoming calls. The Z interface of MSAN_1 and the POTS port of MSAN_2 are in one-to-one mapping and are configured on a 1:1 basis.

Figure 1-121 Principle of IP Z interface extension



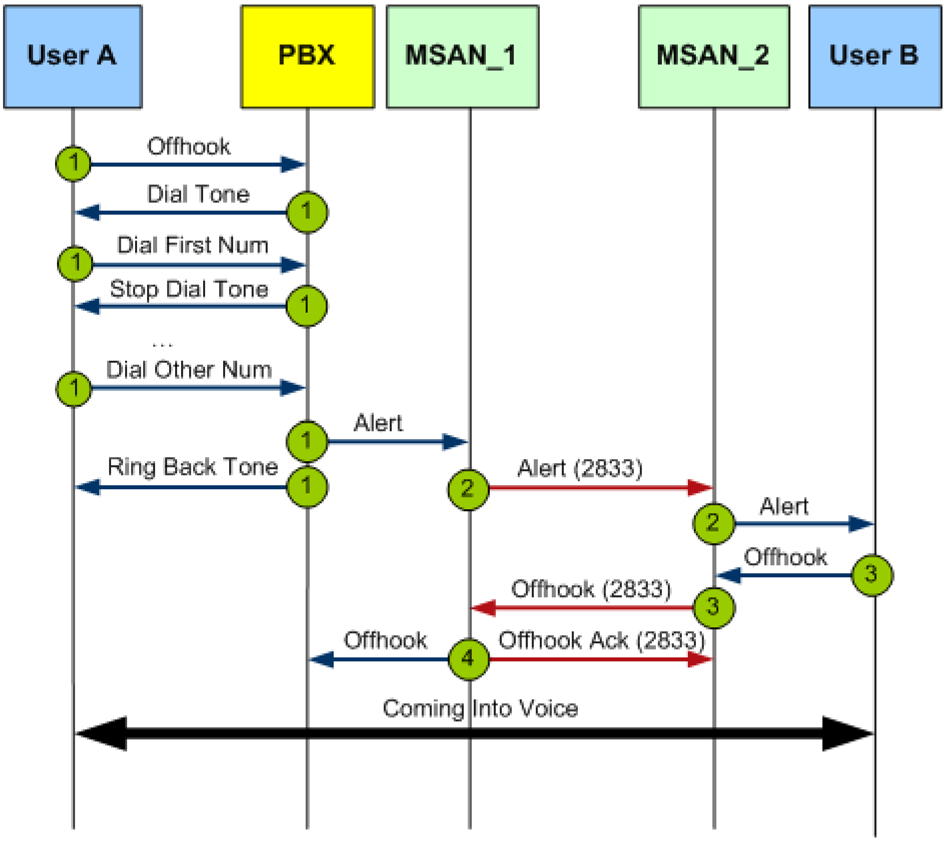
1.14.3 Call Service Flows of IP Z Interface Extension

IP Z interface extension supports many service flows.The following describes the call service flows which include outgoing call service flow for POTS user A, outgoing call service flow for user B with IP Z interface extension, call release flow for POTS user A and call release flow for user B with IP Z interface extension.

Outgoing Call Service Flow for POTS User A

Figure 1-122 shows the outgoing call service flow for POTS user A.

Figure 1-122 Outgoing call service flow for POTS user A



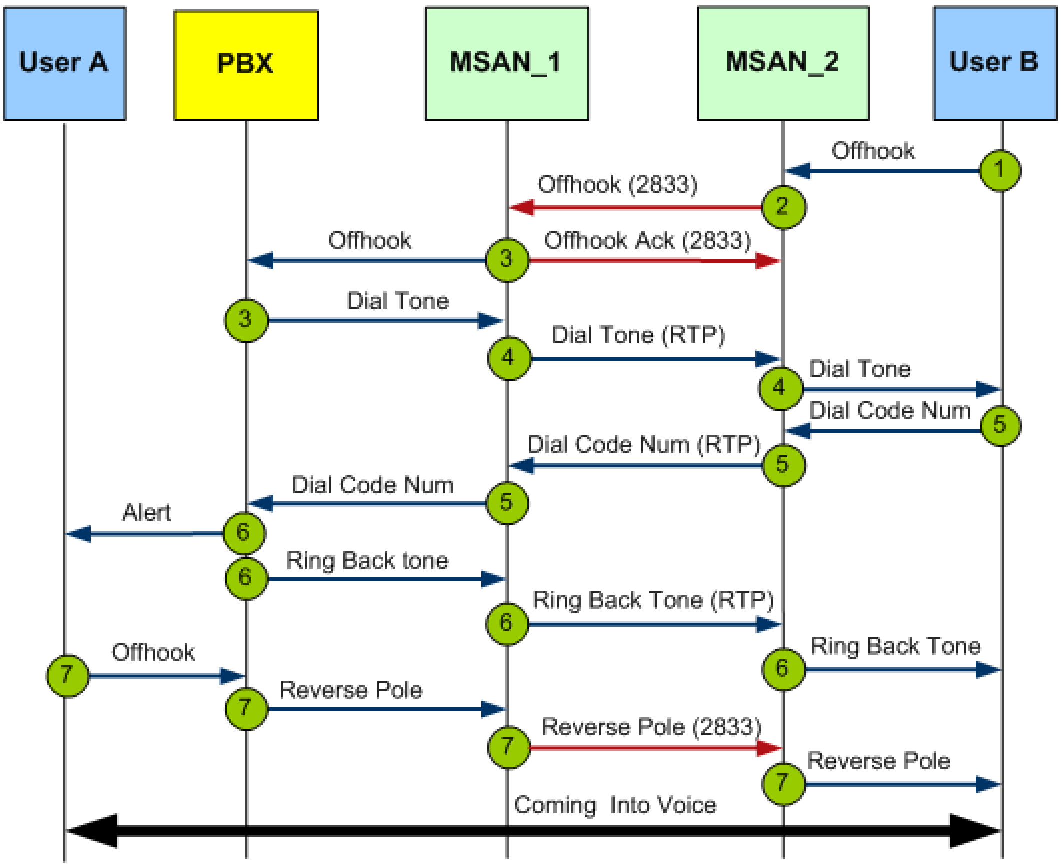
The outgoing call service flow for POTS user A is as follows:

1. User A picks up the phone, the PBX detects the offhook signal of user A and sends the dial tone to user A, and then user A starts to dial the number. After collecting the number, the PBX plays the ringing tone to the called port, and at the same time plays the ringback tone to user A.
2. After detecting the ringing tone, MSAN_1 transmits the ringing tone information to the FXS board of MSAN_2 through an RFC2833 packet. After receiving the RFC2833 packet, MSAN_2 restores the analog signals from the ringing tone signals and plays the analog ringing tones to user B.
3. After hearing the ringing tone, user B picks up the phone. After detecting the offhook signal of user B, the FXS board of MSAN_2 sends the offhook information to MSAN_1 through an RFC2833 packet.
4. After receiving the offhook information, MSAN_1 sends an offhook acknowledge message to MSAN_2 through RFC2833, and at the same time informs the PBX that user B has picked up the phone.
5. After receiving the offhook acknowledge message, the entire speech channel is set up, and the call is established between user A and user B.

Outgoing Call Service Flow for User B with IP Z Interface Extension

Figure 1-123 shows the outgoing call service flow for user B with IP Z interface extension.

Figure 1-123 Outgoing call service flow for user B with IP Z interface extension



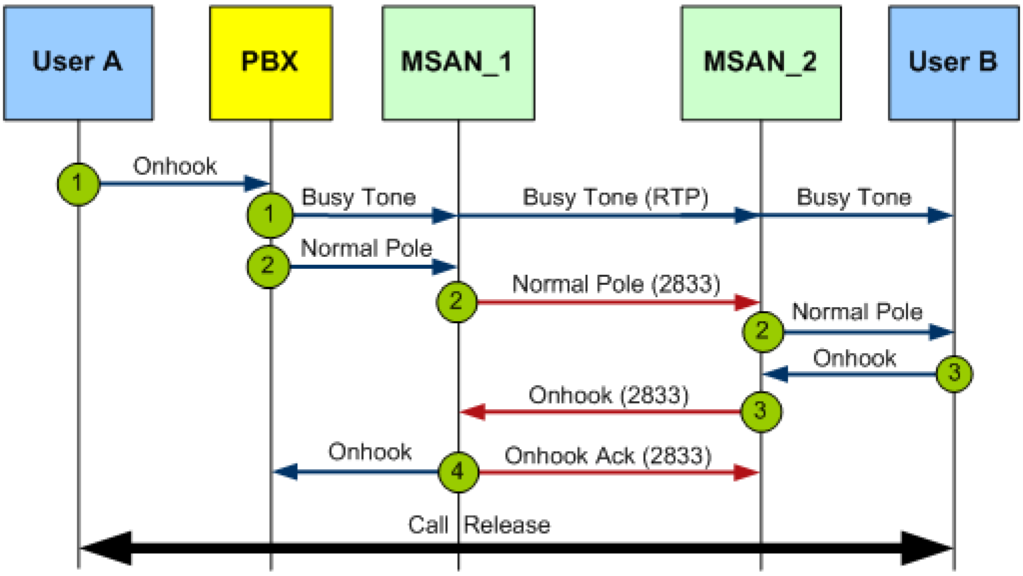
The outgoing call service flow for user B with IP Z interface extension is as follows:

1. User B picks up the phone.
2. MSAN_1 detects the offhook signal of user B and sends the offhook signal through an RFC2833 packet to inform MSAN_2 that user B has picked up the phone.
3. MSAN_2 sends an offhook acknowledgement message to MSAN_1 through RFC2833, and at the same time sends the offhook signal to the PBX. After receiving the offhook signal, the PBX plays the dial tone to MSAN_1.
4. MSAN_1 sends the received dial tone to MSAN_2 through RTP streams, and MSAN_2 sends the dial tone to user B.
5. User B dials the number. The FXS board of MSAN_2 sends the dialed number to MSAN_1 through RTP streams.
6. The PBX collects the number, plays the ringing tone to user A, and at the same time transparently transmits the ringback tone to user B through RTP streams.
7. User A hears the ringing tone and picks up the phone. The PBX detects the offhook signal of user A and sends a polarity reversal signal to MSAN_1. MSAN_1 sends the polarity reversal information to MSAN_2 through an RFC2833 packet. MSAN_2 performs polarity reversal billing on user B. By now, the entire speech channel is set up, and the call is established between user A and user B.

Call Release Flow for POTS User A

Figure 1-124 shows the call release flow for POTS user A.

Figure 1-124 Call release flow for POTS user A



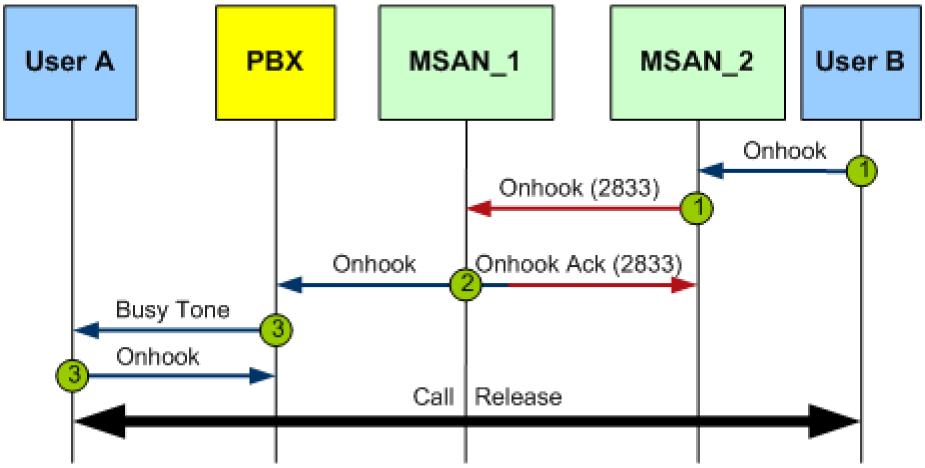
The call release flow for POTS user A is as follows:

1. User A hangs up the phone. The PBX plays the busy tone, and MSAN_1 transparently transmits the busy tone.
2. After user A hangs up the phone, the PBX restores the normal polarity. MSAN_1 detects the polarity restoration message on the port and sends the information to MSAN_2 through an RFC2833 packet. Then, MSAN_2 restores the polarity of user B.
3. User B hears the busy tone and hangs up the phone. MSAN_2 sends the onhook event to MSAN_1 through RFC2833 signaling.
4. When receiving the onhook message, MSAN_1 sends an onhook acknowledgement message to MSAN_2 through RFC2833, and at the same time sends an onhook message to the PBX.
5. After receiving the onhook message, the PBX releases the call.

Call Release Flow for User B with IP Z Interface Extension

Figure 1-125 shows the call release flow for user B with IP Z interface extension.

Figure 1-125 Call release flow for user B with IP Z interface extension



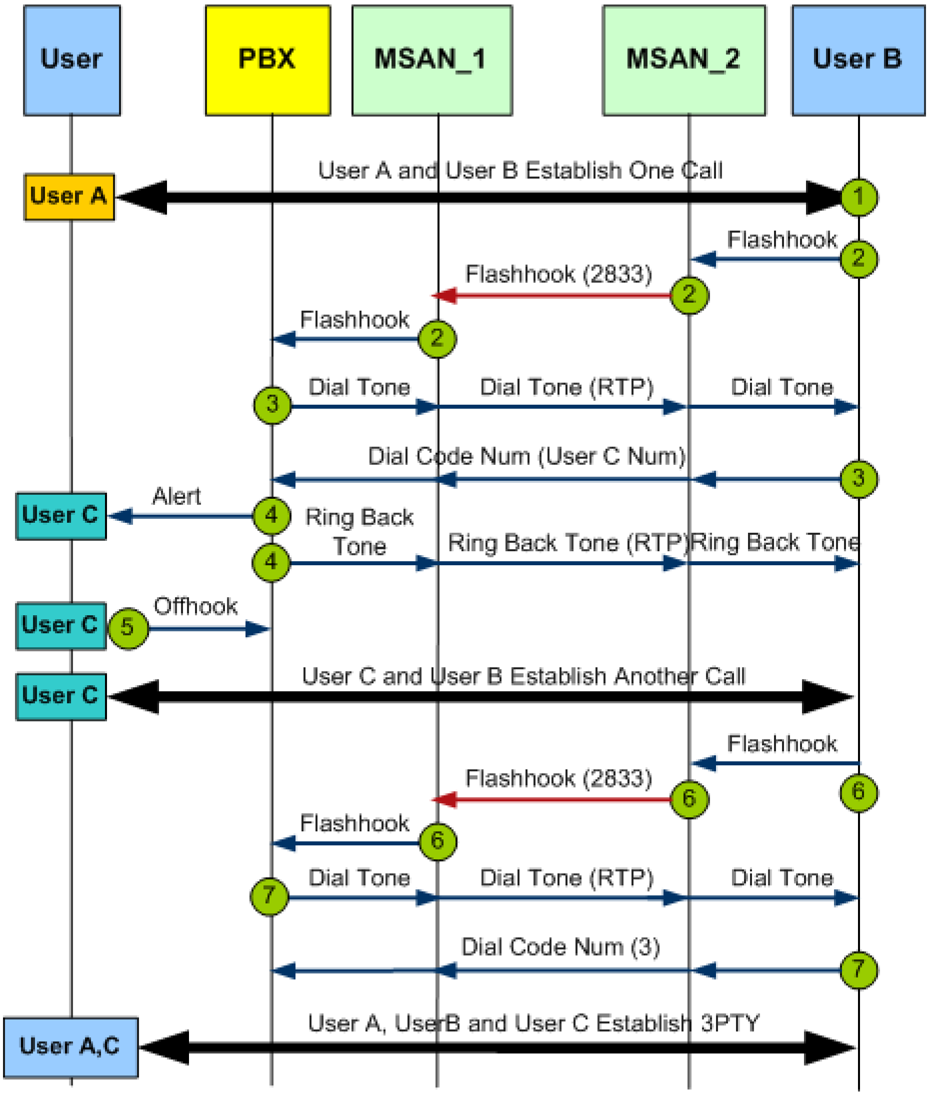
The call release flow for user B is as follows:

1. User B connected to the FXS board hangs up the phone. MSAN_2 detects the local onhook event and sends the event to the FXO board of MSAN_1 through RFC2833.
2. The FXO board sends an onhook acknowledgement to the FXS board through an RFC2833 packet and sends an onhook message to the PBX.
3. After receiving the onhook message, the PBX plays the busy tone to user A. User A hangs up the phone. The call is released.

1.14.4 Carrying Value-added Service Flows of IP Z Interface Extension

The IP Z interface extension feature supports value-added services that include call waiting and three-way calling (3WC). The following uses 3WC as an example to describe how the value-added service is carried using the IP Z interface extension feature. Figure 1-126 shows the service flow.

Figure 1-126 Flow of carrying the 3WC service through IP Z interface extension



The flow of carrying the 3WC service through IP Z interface extension is as follows:

1. A call has been set up between user A and user B.
2. User B needs to communicate with user C and therefore presses the hookflash. MSAN_2 detects the hookflash message, encapsulates the message as an RFC2833 packet, and sends the packet to the FXO board of MSAN_1. MSAN_1 restores the hookflash message from the RFC2833 packet and sends the message to the PBX.
3. After receiving the hookflash message, the PBX transparently transmits the dial tone to user B. After hearing the dial tone, user B starts to dial the number of user C. The dialed number is transparently transmitted to the PBX.
4. After collecting the number, the PBX plays the ringing tone to user C and plays the ringback tone to user B.
5. User C picks up the phone. The call is established between user B and user C.
6. User B presses the hookflash again. MSAN_2 sends the hookflash signal to MSAN_1 through RFC2833. MSAN_1 informs the PBX of user B's hookflash pressing.

7. After receiving the hookflash message, the PBX transparently transmits the dial tone to user B. After hearing the dial tone, user B starts to dial the DTMF number 3. The dialed number is transparently transmitted to the PBX. By now, the 3WC is established between users A, B, and C.

NOTE

During the time from user B pressing the hookflash for the first time to user B pressing the hookflash for the second time and dialing the DTMF number 3, user A is in the call waiting state.

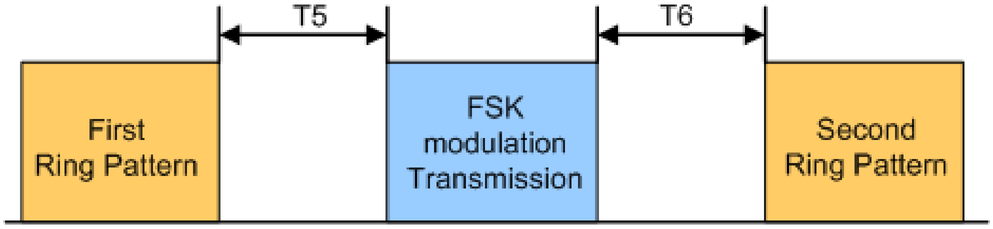
1.14.5 Ringing and CLIP Services for IP Z Interface Extension Feature

When a PSTN network is restructured into an IP network, the ringing and calling line identification presentation (CLIP) services for IP Z interface extension users need some implementation changes accordingly. This topic describes the context and implementation changes for the FXO port to support the ringing and CLIP services for IP Z interface extension users serving as the called parties.

Context

- Figure 1-127 illustrates the intervals of the ringing and CLIP signals in the on-hook process specified in ETSI EN 300 659-1.

  Figure 1-127 Intervals of the ringing and CLIP signals in the on-hook process (ETSI EN 300 659-1)



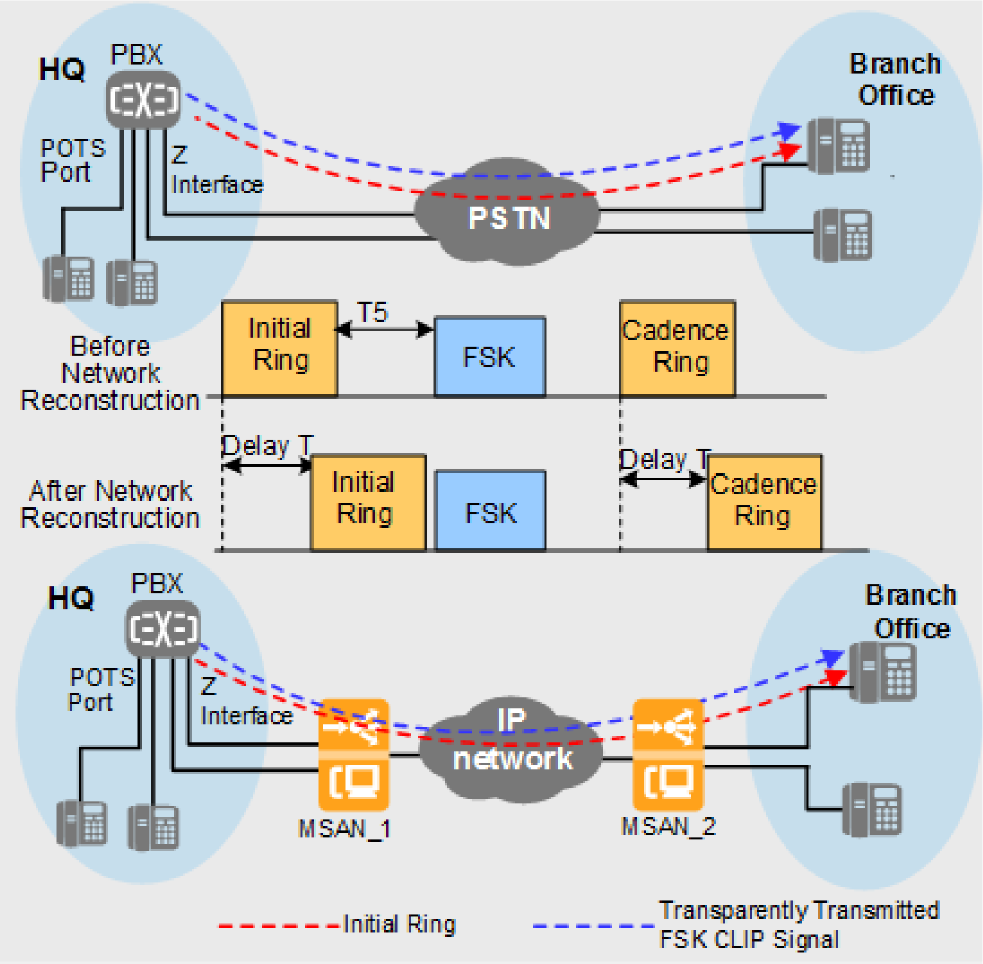
  According to the preceding figure, the interval between the first ring pattern signal and the FSK modulation transmission signal (CLIP signal) is T5 (500 ms ≤ T5 ≤ 2000 ms); the interval between the FSK modulation transmission signal and the second ring pattern signal is T6.

  The **First Ring** parameter in the protocol is the **Initial Ring** parameter of the MSAN, and the **Second Ring** parameter in the protocol is the **Cadence Ring** parameter of the MSAN.

- Compared with a PSTN network, an IP network needs more time (delay T) to send ringing signals of a Z interface extension call from the PBX to the telephone, because the ringing signals require processing by the MSAN on an IP network, as shown in Figure 1-128.

  However, both PSTN and IP networks transparently transmit FSK modulation transmission signals from the PBX to the telephone and thereby spend the same time. As a result, an IP network has a shorter T5 (interval between the first ring pattern signal and the FSK modulation transmission signal) than a PSTN network does.

Figure 1-128 T5 difference between a PSTN network and an IP network



Theoretically, when a call to an IP Z interface extension user is ended before it is answered, the PBX hook on then stop sending a ringing message to the MSAN_1 and then the MSAN_1 detect ringing missing message to send a ringing stop message to the MSAN_2. However, the ringing stop message is also delayed due to MSAN processing.

If the delay is not properly handled, the CLIP service may be affected, and the telephone may still ring after a call is ended.

Implementation Changes

- According to the preceding analysis, delay T must be shorter than T5 to ensure that the CLIP service is normal.

  Delay T = Ringing signal detection time of the FXO board + Time used by Z interface extension (about 150 ms, including codec, jitter buffer, and other message processing time)

  The ringing signal detection time of the FXO board can be configured using the **min-ontime** parameter of the **fxoport attribute set** command.

  The value of the **min-ontime** parameter must meet the following requirement: **min-ontime** < T5-150 ms.

NOTE

For the value of T5, see ETSI EN 300 659-1.

- The FXO board must spend as little time as possible in detecting ringing missing message to avoid the undesired telephone ringing after a call is ended.

  The ringing missing signal detection time of the FXO board can be configured using the **max-offtime** parameter of the **fxoport attribute set** command.

NOTE

When you configure the **max-offtime** parameter, take the ringing break-make ratio into consideration. If the **max-offtime** parameter is set to a time shorter than the ringing break, ringing signals cannot be detected.

1.15 Key Techniques for Improving Voice Service Quality

Voice service quality is the biggest challenge faced by the IP telephony technology. IP telephony service has a higher requirements on real-time transmission of IP packets. If IP packets are lost, or transmission delay or jitter is introduced due to transmission errors or network congestion, subscribers hear noises during calls, and even more, ongoing calls may be interrupted. The VoIP technology provides a series of technologies, such as codec, echo cancellation (EC), and voice activity detector (VAD) to improve the voice quality.

1.15.1 Codec and Packetization Duration

Introduction

Codec is a key technology of voice services. Coding means that the DSP encodes the TDM-based voice data, assembles the data into packets, and then sends the packets to the IP network. Decoding means that the DSP decodes the voice packets received from the IP network and plays the voice to the TDM side.

Frequently-used codec types are G.711A, G.711Mu, G.729a, G.729b, G.726, G.723.1Low, and G.723.1High. G.711A and G.711Mu are lossless coding schemes. G.729, G.723.1Low, and G.723.1High are lossy compressed coding schemes. The compressed coding schemes require less bandwidth, but the voice quality is poor and the delay is large. (G.711 delivers the best voice quality but requires a bandwidth of 64 kbit/s. G.723 requires less bandwidth but the voice quality is less satisfying.)

PTime is the interval at which the DSP assembles the voice data into packets. It varies according to the codec type. Table 1-15 lists the codec types.

Table 1-15 Codec list

Codec Type	Coding Rate (kbit/s)	PTime
G.711A/G.711Mu	64	10 ms, 20 ms, 30 ms
G.729a/G.729b	8	10 ms, 20 ms, 30 ms
G.726	16/24/32/40	10 ms, 20 ms, 30 ms
G.723.1	5.3/6.3	30 ms, 60 ms

Standards and Protocols Compliance

ITU-T G.711, ITU-T G.729, ITU-T G.726 and ITU-T G.723

1.15.2 EC

Introduction

Echo canceller (EC) is to cancel echo during calls. Echo is classified into the acoustic echo and electrical echo.

- Acoustic echo

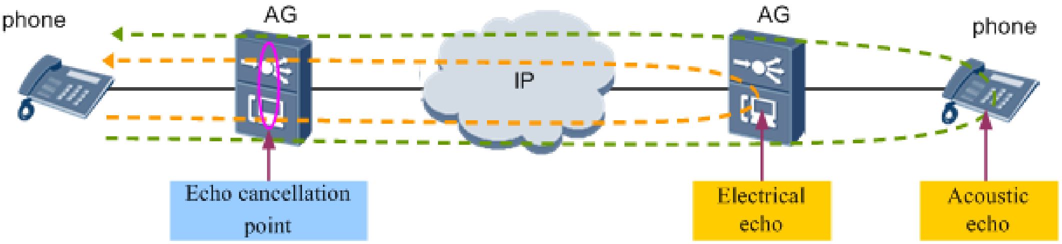
  Acoustic echo refers to the echo reflected by an obstacle when the voice encounters the obstacle in the transmission path. For example, if you place the phone at one side and speak at the other side, you can hear your own voice. This is because the voice is transmitted through the table and reflected from the collector to the receiver of the phone. Currently, the VoIP DSP chip does not support cancellation of the acoustic echo because it cannot distinguish the normal voice from the acoustic echo.

- Electrical echo

  Electrical echo is generated by the 2-wire/4-wire converter on the service board, because the impedance matching is not ideal on the 2-wire/4-wire converter. EC generally refers to the cancellation of the electrical echo.

Figure 1-129 shows how the electrical echo is generated.

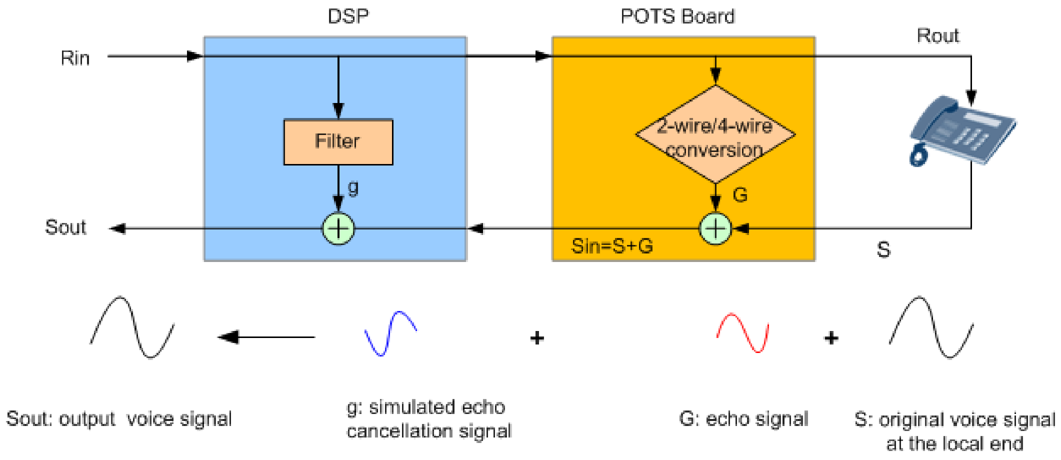
Figure 1-129 EC



In the PSTN network, owing to the small delay, the voice and the echo reach the ears of the speaker almost at the same time. Therefore, the echo can hardly be perceived. On the VoIP network, owing to the large delay, the echo reaches the ears some time after the voice is heard. Therefore, the echo can be easily perceived. As described in ITU-T G.131 and ITU-T G.161, the echo can be perceived when the echo delay exceeds 25 ms.

Figure 1-130 shows how the EC is implemented.

Figure 1-130 Implementation of the EC function


DSP
POTS Board
Rin
Rout
Filter
g
2-wire/4-wire conversion
G
Sout
Sin=S+G
S
+
+
Sout: output voice signal
g: simulated echo cancellation signal
G: echo signal
S: original voice signal at the local end

Rin is the voice received from the remote end. **Rin** is the input of the wave filter and the output of the wave filter is the simulated echo cancellation signal **g**. During the 2-wire/4-wire conversion, echo **G** is generated based on **Rin**. **S** is the original voice signal at the local end, that is, the voice received by the local receiver. The local-end voice signal **S** is overlaid with the echo cancellation signal **G**, resulting in the input signal of the EC, **Sin**. The EC removes the simulated echo **g** from the input signal **Sin** to obtain the output signal **Sout**.

Sin = S + G

Sout = Sin - g = S + G - g

G ≈ g

Therefore, Sout ≈ S

Reference Standards and Protocols

ITU-T G.168, ITU-T G.131, and ITU-T G.161

1.15.3 Non-Linear Processor

Introduction

Owing to various reasons, the EC cannot cancel all the echoes. To improve the EC performance, a non-linear processing (NLP) is performed on the remaining echoes when the power of the remaining echoes is lower than a preset value. This can further reduce the power of the remaining echoes. A simple method is to replace the remaining echoes with the silence when the power of the remaining echoes is lower than the threshold.

Specifications

The NLP function can be enabled or disabled by configuring the DSP profile on a port. If the DSP profile is not configured, the system automatically enables or disables the NLP function according to the service mode. Specifically, for the voice service, the system enables the NLP function; for the fax or modem service, the system disables the NLP function.

Impact

The NLP function must be disabled in the case of FoIP or MoIP.

Reference Standards and Protocols

ITU-T G0.168, ITU-T G0.131, and ITU-T G0.161

1.15.4 VAD/CNG

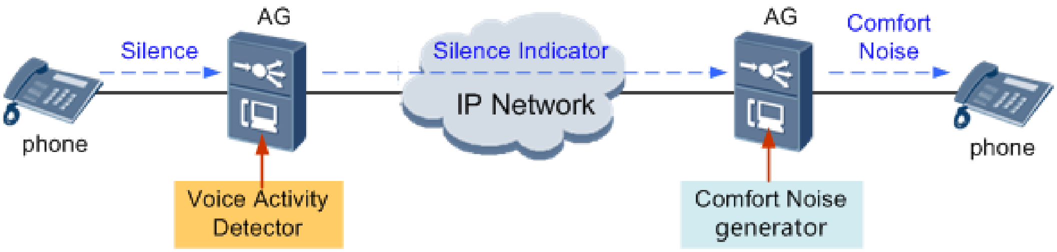
Introduction

According to statistics, silent duration exceeds 50% of the total session duration. If data is still transmitting in common packetization mode during the silent period, network bandwidth resources will be wasted. The voice activity detector (VAD) and comfort noise generator (CNG) significantly reduce network bandwidth usage to ease the insufficiency of network resources.

- VAD: The VAD identifies voice and silent durations based on signal energy. After detecting silence, the Tx end only sends mute indication packets instead of voice IP packets, lowering occupied bandwidth.
- CNG: To avoid long time silence during the mute period that may discomfort the user, the Rx end needs to generate comfort noises during the mute period according to the mute indication sent by the Tx end.

VAD and CNG are always used together. VAD is used on the Tx end, and CNG is used on the Rx end. For example, when VAD is enabled, the DSP packetizes RTP packets and sends the packets to the remote end only when it detects voice signals. In the case of silence, the DSP does not send RTP packets to the IP side. The DSP sends a silence ID (SID) to inform the remote end only when the background noise changes. The remote DSP then generates background noises according to the information carried in the SID. Figure 1-131 shows the implementation process.

Figure 1-131 Networking for VAD and CNG



Specifications

ITU-T G.711 and ITU-T G.729

1.15.5 PLC

Introduction

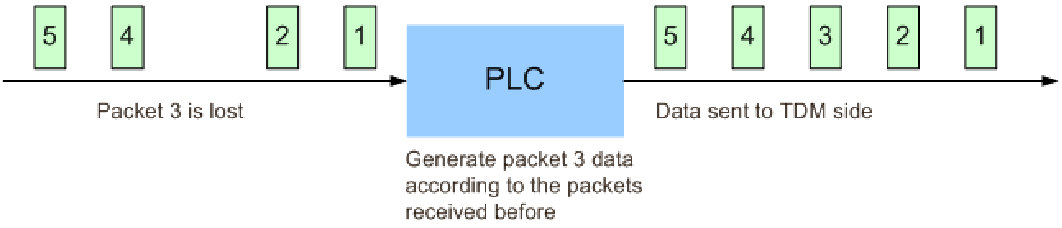
When a network or a device loses packets, the voice quality deteriorates. In practice, packet loss is inevitable. If the packet loss concealment (PLC) is enabled to compensate the signals, however, the impact of packet loss on the voice quality is reduced and the success rates of FoIP and MoIP services increases in the case of packet loss. Voice IP data is transmitted using Real-Time Transport Protocol (RTP). The DSP chip detects the sequence number of received RTP packets. If the DSP chip detects packet loss, it automatically constructs voice data based on a configured compensation algorithm and sends the data to the TDM side.

Three compensation modes are available:

- Compensate the lost packet with the silence.
- Compensate the lost packet with the previous packet.
- Compensate the lost packet with a similar packet that is calculated based on the energies of the previous packet and the subsequent packet (as described in G.711 Appendix I).

The third mode consumes the most DSP resources, but improves the voice quality in the most satisfying manner. The first mode consumes the least DSP resources, but improves the voice quality in the least satisfying manner. Figure 1-132 shows the packet loss compensation using the previous packet.

Figure 1-132 Packet loss compensation using the previous packet


5
4
2
1
Packet 3 is lost
PLC
Generate packet 3 data according to the packets received before
5
4
3
2
1
Data sent to TDM side

NOTE
When the IP network quality is poor, if the number of consecutively lost RTP packets is greater than 2, the voice quality is still poor even if the PLC is enabled.

Reference Standards and Protocols

G.711 Appendix I

1.15.6 JB

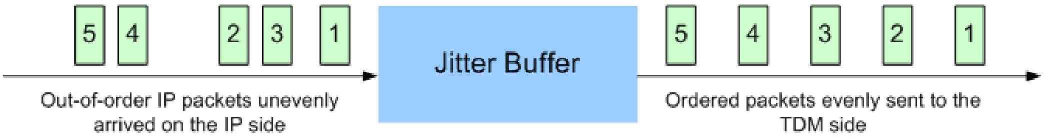
Introduction

The transmission quality on the IP network is not guaranteed. The interval at which packets are received from the remote end is not even, and the sequence of packets received may be different from the sequence that these packets are sent. As a result, the voice quality is degraded. Therefore, the jitter buffer (JB) is introduced to eliminate the jitter of the IP network. The basic idea of JB is that delay is introduced so that uneven and disordered RTP packets can be sequenced in the buffer and then sent to the TDM side. Figure 1-133 shows the JB implementation method.

The JB is classified into the dynamic JB and the static JB.

- Dynamic JB: The buffer depth can be automatically adjusted based on network jitter conditions, so that the introduced delay is appropriate to process the jitter. The dynamic JB mainly applies to voice services.
- Static JB: The buffer depth is fixed and cannot be changed based on network jitter conditions. The static JB mainly applies to data services, such as fax over IP (FoIP) and modem over IP (MoIP). This is because the modification of buffer depth may lead to packet loss, which has great adverse impact on data services.

Figure 1-133 JB implementation method


5
4
2
3
1
Out-of-order IP packets unevenly arrived on the IP side
Jitter Buffer
5
4
3
2
1
Ordered packets evenly sent to the TDM side

Specifications

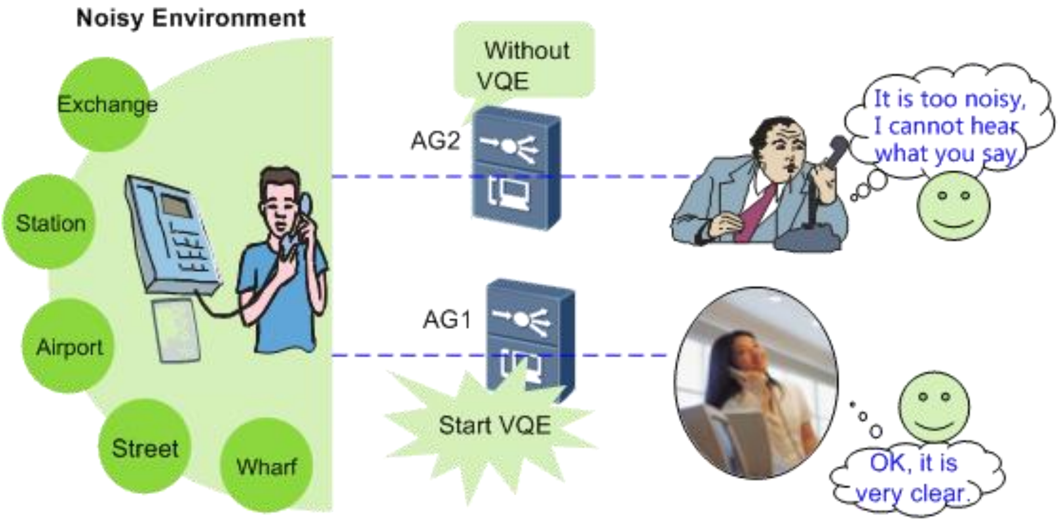
The dynamic JB and the static JB are supported. The adjustable range of the JB depth is 0 ms to 135 ms.

1.15.7 VQE

Introduction

The Voice Quality Enhanced (VQE) feature applies to voice services in the noisy public areas, such as the roads, docks, scenic spots, and bus stations. Deployment of VQE in these areas can improve the voice quality and user experience. Figure 1-134 shows application scenarios and advantage of the VQE.

Figure 1-134 Application scenarios and advantage of the VQE



The VQE consists of two functions, automatic gain control (AGC) and spectral noise suppression (SNS).

- AGC: A target gain energy value is configured so that the output gain can be automatically adjusted during the VoIP communication, ensuring that subscriber can still hear voices in a noisy environment. AGC ensures smooth energy adjustment and prevents adverse impact brought by abrupt energy changes.
- SNS: A target noise suppression value is configured. After detecting that the noise energy value is greater than the target value, AG reduces the noise energy value to enable subscribers to feel comfortable during VoIP communication.

Specifications

The AG only supports AGC. The VQE feature is configured based on user ports and takes effect for calls initiated after the VQE is enabled.

1.15.8 Fax/Modem Quality Enhancement

Overview

After the IP network takes the place of the PSTN network, the use of fax and modem on the VoIP network becomes more and more popular. Therefore, the AG is required to provide applications similar to those of the PSTN network. Currently, the Voice Band Data (VBD) transparent transmission is adopted by the medium gateway (MG) in the application of the fax and modem. Transparent transmission, however, relies heavily on the bearer network and deterioration of the network quality may lead to service failures.

The fax/modem quality enhancement feature consists of the RFC2198 intelligent startup function and the packetization at the interval of 10 ms. After the fax/modem quality enhancement feature is enabled, the RFC2198 function and the packetization at the interval of 10 ms are automatically started. The following table provides more details.

Problem	Solution	Description
Packet loss	RFC 2198	The RFC 2198 standard uses the data stream redundancy mechanism to prevent the packet loss of the network from degrading the service quality. When the average consecutive packet loss ratio is low, the receiver can reassemble and restore the lost packet based on the redundant packets in the later received packets. The audio redundancy mechanism described in RF2198 can be used to restore the events lost in the packets, while the mode described in RFC2833 can be used to process the DTMF signals transmitted through the RTP packets.
Network delay	10-ms packetization	Information carried by packets assembled every 10 ms is less the information carried by packets assembled every 20 ms. Therefore, in case of packet loss, the packetization at the interval of 10 ms causes less impact on services that the packetization at the interval of 20 ms. Using this technology, the device automatically detects fax and modem signals and switches the 20-ms packetization interval (intended for voice services) to the 10-ms packetization interval, thereby reducing the network delay of fax and modem data transmission.

The enhanced quality feature of the fax and modem is mainly used to improve the put-through rate and online duration of the fax and modem services. For example, if POS terminals are connected to a modem in a shopping mall or a bank, the fax/modem quality enhancement feature can be used to improve the stability and online duration of the Modem, thus preventing the disconnection caused by the poor network quality, as shown in Figure 1-135.

Figure 1-135 Enhanced Modem

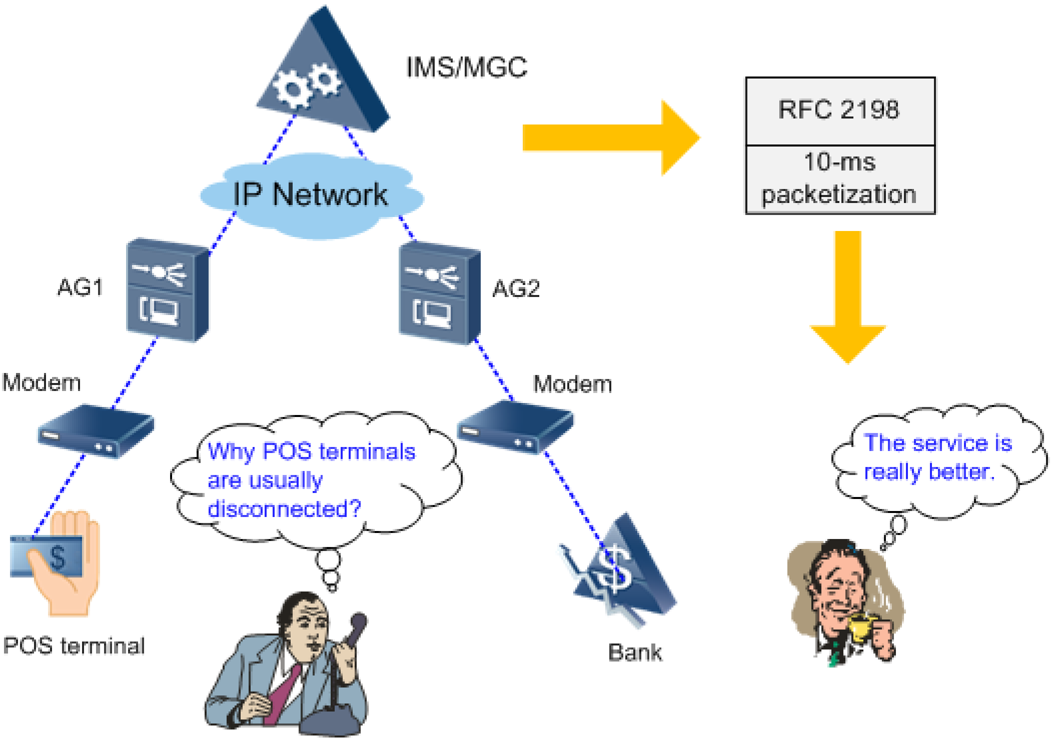


Table 1-16 lists the test data recorded before and after the quality enhancement feature is enabled. The data shows that stability of services improves substantially after the quality enhancement feature is enabled.

Table 1-16 Test data before and after the quality enhancement

Modem	PTime	Network Packet Loss Ratio (Random)	Online Duration
T336CX	20 ms	0.10%	22 hours
		0.50%	1.5 hours
		1.00%	Unavailable
		1.00% (rfc2198)	12.5 hours
	10 ms	0.10%	> 24 hours
		0.50%	22.5 hours
		1.00%	5.5 hours

Modem	PTime	Network Packet Loss Ratio (Random)	Online Duration
		1.00% (rfc2198)	24 hours

Specifications

The fax/modem quality enhancement is supported.

1.16 Voice Service Maintenance and Diagnosis

The maintenance and diagnosis features of voice services include these features such as the loop-line test, circuit test, call emulation , POTS port loop test, VBD fault diagnosis, Real-time Transport Control Protocol (RTCP) statistics and so on.

1.16.1 Call Emulation Test

A call emulation test emulates call functions to verify data configuration for the voice service. The call emulation test can also be used to locate voice service faults.

1.16.1.1 Introduction to the Call Emulation Test

Definition

In a call emulation test, the device emulates the call function of a voice user. It is used to test the services on the POTS user port. A call emulation test includes:

- **Calling party emulation test**: The POTS user port on the device functions as the calling party. In this test, a test engineer acting as a called party is required.
- **Called party emulation test**: The POTS user port on the device functions as the called party. In this test, a test engineer acting as a calling party is required.
- **Calling and called party emulation test**: Two POTS user ports function as the calling and called parties. The test does not require manual operations. Specifically, the device automatically performs the process from calling party off-hook to called party off-hook to set up a call between the two parties and stops the test after the call hold time expires.

Application Scenarios

A call emulation test can be used in the following scenarios:

- Acceptance test during a new deployment: The software and hardware functions, including service configurations, of the device need to be verified after the device is installed. The verification ensures follow-up service provisioning.

  Traditionally, the engineer goes to the device installation site, makes cables, connects a test phone set to the ONT, uses the test phone set as a caller or callee, and verifies basic voice services.
- Fault locating in the OAM phase: After the device enters the OAM phase, it is usually necessary to test basic voice services in order to locate a fault. In the access network, however, a large number of devices are installed in complicated environment,

geographically dispersed, and remotely located. It is inconvenient and costly either to test a newly installed device or locate faults.

Call emulation tests can be conducted remotely. In a call emulation test, the test engineer does not need to prepare cables on the device installation site, connect test terminals to the device, or perform dialup tests on site. Instead, the test engineer enables the call emulation function in the maintenance center through the command line interface (CLI) or network management system (NMS) and uses a test phone set in the central office (CO) to make calls to the emulation port on the device. In this way, the test engineer can verify the data configurations and basic service functions.

Benefits

- The call emulation feature can be used to remotely verify and accept services and locate faults, which greatly reduces the operating expense (OPEX) for carriers.
- The call emulation feature shortens fault location time, which significantly improves the fault locating efficiency.

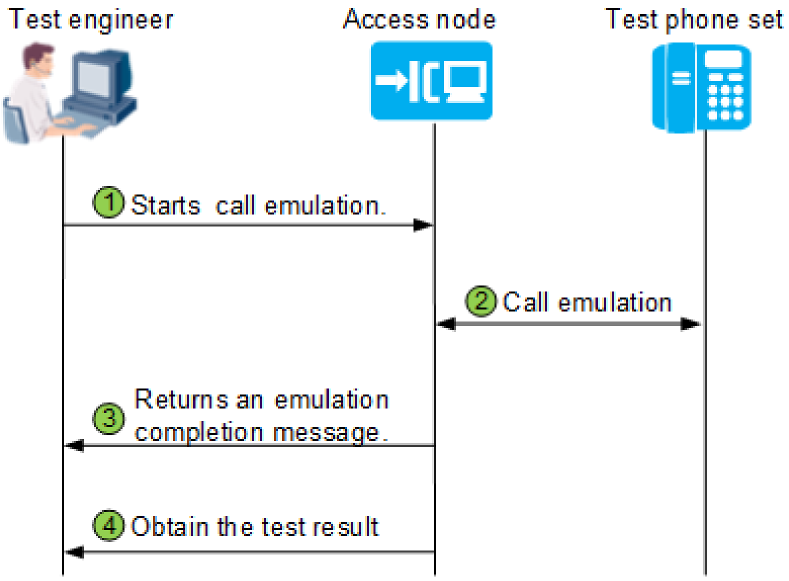
1.16.1.2 Principles of the Call Emulation Test

In a call emulation test, the system emulates the actions of a user port to achieve the emulation of the calling party and called party. The actions of a user port include off-hook, on-hook, number dialing, and ringing detection. Figure 1-136 shows the principles of a call emulation test. The test phone set, placed at the central office (CO), is used to perform a call emulation test with a configured port on the access node. This test method remotely checks whether the voice service on the device is functional.

NOTE

For a calling and called party emulation test, no phone set is required. Both the calling and called parties are emulated by the POTS ports on the access node.

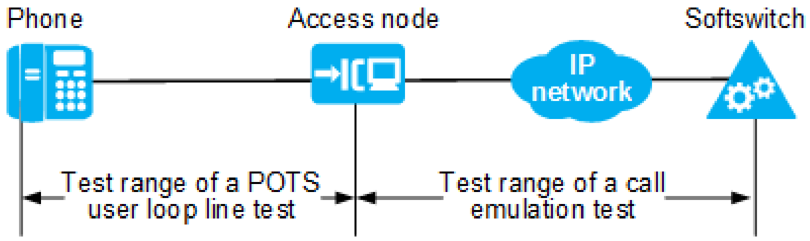
Figure 1-136 Principles of a call emulation test



Networking Application

Figure 1-137 shows the example network of a call emulation test.

Figure 1-137 Example network of a call emulation test


Phone
Access node
Softswitch
IP network
Test range of a POTS user loop line test
Test range of a call emulation test

A call emulation test checks whether the voice service is functional only on the network side of the device.

- If a conversation channel cannot be established during a call emulation test, check whether the network data configurations are correct.
- If the conversation channel is established but the voice is not clear during a call emulation test, check whether the cables are connected properly.

If test engineers need to check whether the voice service is functional on the user side, they can:

- Perform a POTS user loop line test to check whether the line between the device and the user phone set is functional. For details, see POTS User Loop Line Test.
- Perform a POTS user circuit test to check whether the POTS board on the device is functional. For details, see POTS User Circuit Test.

Calling Party Emulation Test

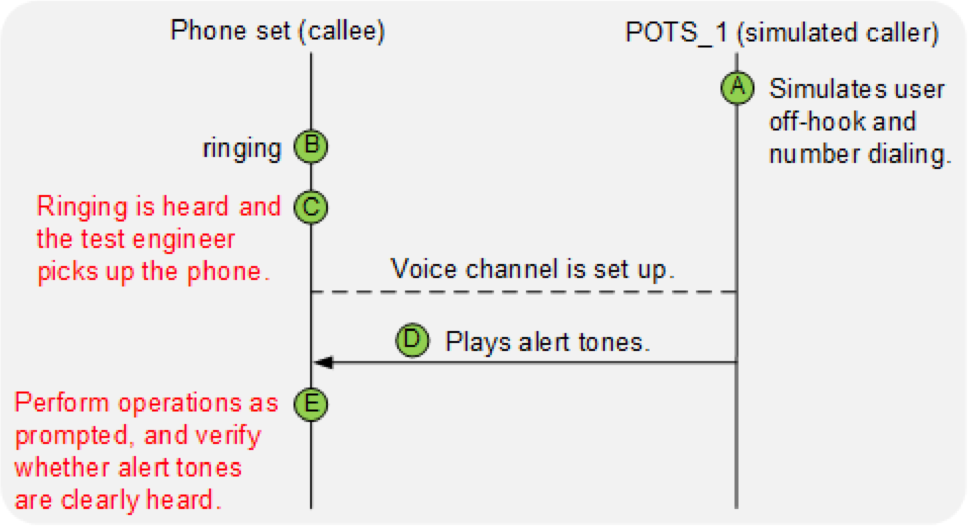
In a calling party emulation test, the device port emulates user off-hook, number dialing, communication, and on-hook.

1. Start a calling party emulation test.

   On the access node, the test engineer sets the POTS_1 port as the calling party emulation port, configures the phone number to be dialed, and starts the calling party emulation test. If the function of playing alert tones upon the test beginning is enabled, alert tones are played after the test is started.

2. Initiate the calling party emulation test.

Figure 1-138 Interaction between the POTS_1 port and phone set during a calling party emulation test


Phone set (callee)
POTS_1 (simulated caller)
A
Simulates user off-hook and number dialing.
ringing
B
C
Ringing is heard and the test engineer picks up the phone.
Voice channel is set up.
D
Plays alert tones.
E
Perform operations as prompted, and verify whether alert tones are clearly heard.

NOTE

The information marked red in the following figure indicates the operations that need to be performed by the test engineer on the test phone set at the CO.

a. The calling party emulation test is started on the access node and the calling party emulation port automatically emulates user off-hook. After detecting the dial tone, the port emulates number dialing.

b. The called party (whose number is automatically dialed by the calling party emulation port) waits for the phone to ring. If the phone rings, the signaling channel is functional and the configured data is correct. If the phone does not ring, the test engineer needs to check service data (such as route, VLAN, and core-network data), troubleshoot the voice fault if any, and perform the test again.

c. The called party picks up the phone. The call is set up.

d. The calling party emulation port plays announcements for the called party.

e. The test engineer checks whether the announcements are clearly heard. After hearing the announcements, the test engineer presses the specified verification number (a matched DTMF number, the asterisk key (*) by default), indicating that the media channel is functional. The test result is "Test Succeed."

NOTE

If the function of playing alert tones based on the DTMF matching result is enabled, the system plays the alert tones after the test engineer presses the DTMF number. The alert tones include the DTMF number matching success alert tone and DTMF number matching failure alert tone.

3. Obtain the test result. If the test fails, the system outputs the failure causes, with which the test engineer can identify the possible cause of the fault.

   Table 1-17 lists the test results.

Called Party Emulation Test

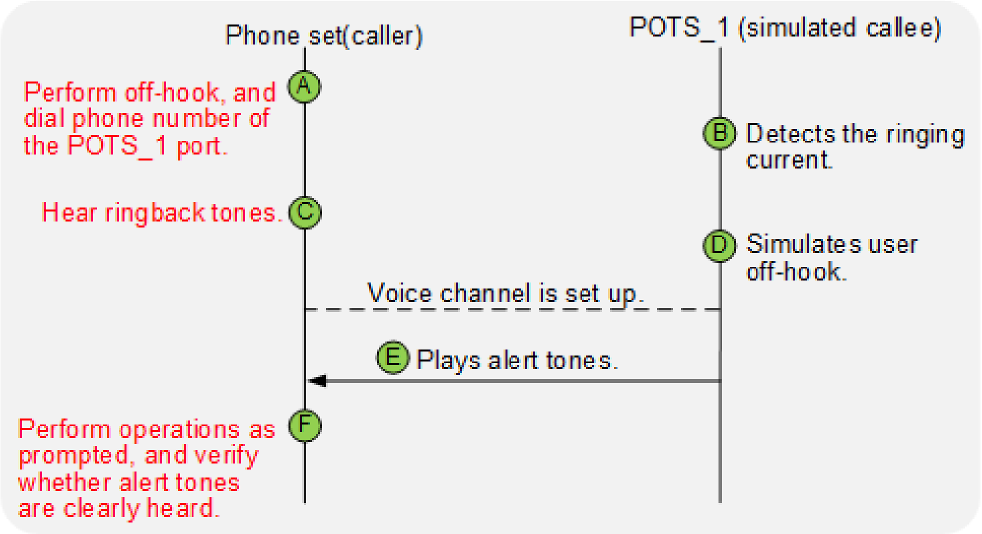
In a called party emulation test, the POTS port on the access node emulates a called party. The called party emulation port automatically emulates user off-hook after detecting the ringing current.

1. Start a called party emulation test.

   On the access node, the test engineer sets the POTS_1 port as the called party emulation port and starts the called party emulation test. If the function of playing alert tones upon the test beginning is enabled, alert tones are played after the test is started.

2. Initiate the called party emulation test.

   Figure 1-139 Interaction between the POTS_1 port and phone set during a called party emulation test


Phone set(caller)
POTS_1 (simulated callee)
Perform off-hook, and dial phone number of the POTS_1 port.
A
B
Detects the ringing current.
Hear ringback tones.
C
D
Simulates user off-hook.
Voice channel is set up.
E
Plays alert tones.
F
Perform operations as prompted, and verify whether alert tones are clearly heard.

NOTE

The information marked red in the following figure indicates the operations that need to be performed by the test engineer on the test phone set at the CO.

   a. The test engineer picks up the phone, hears the dial tone, and dials the number of the POTS_1 port.
   b. If the called party emulation port of the device detects the ringing current, the configured user data is correct. If the called party emulation port does not detect the ringing current, the test engineer needs to check service data (such as route, VLAN, and core-network data), troubleshoot the voice fault if any, and perform the test again.
   c. If the calling party hears ringback tones before the called party picks up the phone (in this test, off-hook is automatically emulated by the called party emulation port), the signaling channel is functional. Otherwise, the test engineer needs to verify the configured service data and perform the test again.
   d. The called party emulation port emulates off-hook. The call is set up.
   e. The called party emulation port plays announcements for the calling party.
   f. The test engineer checks whether the announcements are clearly heard. After hearing the announcements, the test engineer presses the specified verification number (a matched DTMF number, the asterisk key (*) by default), indicating that the media channel is functional. The test result is "Test Succeed."

NOTE

If the function of playing alert tones based on the DTMF matching result is enabled, the system plays the alert tones after the test engineer presses the DTMF number. The alert tones include the DTMF number matching success alert tone and DTMF number matching failure alert tone.

3. Obtain the test result. If the test fails, the system outputs the failure causes, with which the test engineer can identify the possible cause of the fault.

   Table 1-18 lists the test results.

Calling and Called Party Emulation Test

In a calling and called party emulation test, two POTS ports emulate the calling party and called party respectively. The calling party emulation port emulates the actions of the calling party, while the called party emulation port emulates the actions of the called party. These two ports automatically emulate calling party off-hook, number dialing, called party off-hook (after detecting the ringing current), mutual DTMF number sending for media channel verification, and on-hook after the verification.

NOTE

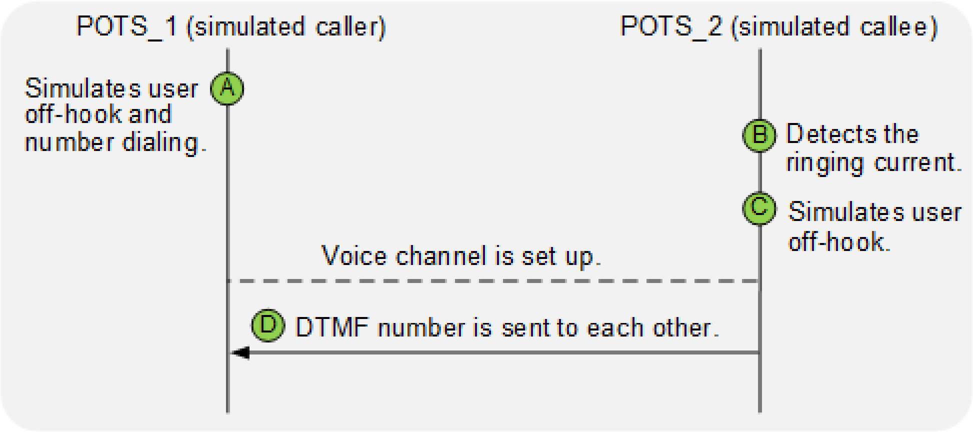
A calling and called party emulation test does not require any manual operation, which greatly improves the test efficiency. In this test, the device automatically verifies whether the media channel is functional. However, the device's sensitivity to media streams may vary from the human ears' sensitivity to media streams. Therefore, the call quality cannot be verified.

1. Start a calling and called party emulation test.

   On the access node, the test engineer sets the POTS_1 port as the calling party emulation port and the POTS_2 port as the called party emulation port, configures the phone number to be dialed, and starts the calling and called party emulation test.

2. Initiate a calling and called party emulation test.

   Figure 1-140 Interaction between the POTS_1 and POTS_2 ports during a calling and called party emulation test


POTS_1 (simulated caller)
POTS_2 (simulated callee)
A
Simulates user off-hook and number dialing.
B
Detects the ringing current.
C
Simulates user off-hook.
Voice channel is set up.
D
DTMF number is sent to each other.

   a. The calling and called party emulation test is started on the access node and the POTS_1 port automatically emulates user off-hook. After detecting the dial tone, the port emulates number dialing.

   b. The POTS_2 port (whose number is automatically dialed by the POTS_1 port) waits and checks whether the ringing current can be detected. If the POTS_2 port detects the ringing current, the signaling channel is functional and the configured user data is correct. If the POTS_2 port does not detect the ringing current, the test engineer needs to check service data (such as route, VLAN, and core-network data), troubleshoot the voice fault if any, and perform the test again.

c. The POTS_2 port emulates off-hook. The call is set up.

d. The POTS_1 and POTS_2 ports send the DTMF number to each other to verify whether the media channel is functional. If the DTMF numbers sent by POTS_1 and POTS_2 ports are correct, the media channel is functional. The test result is "Test Succeed."

3. Obtain the test result. If the test fails, the system outputs the failure causes, with which the test engineer can identify the possible cause of the fault.

   Table 1-17 and Table 1-18 list the test results.

Test Results

Table 1-17 Results of a calling party emulation test

Test Result	Description
Test Succeed	The test is successful. The media channel of the test port is functional.
Test Failed: No dialing tone is played when the calling party dials a number	The calling party emulation port does not detect the dial tone. Possible causes are as follows: • The calling party emulation port does not detect the off-hook signal. • The off-hook signal is not reported. • No dial tone is issued after the off-hook signal is reported.
Test Failed: Busy tone is played after the calling party picks up the phone off the hook	The calling party emulation port detects the busy tone. Possible causes: The calling party emulation port does not subscribe to the POTS services or the digital signal processor (DSP) is faulty.
Test Failed: Busy tone is played when the calling party dials a number	The calling party emulation port detects the busy tone during number dialing. Possible cause: The number that the calling party emulation port dials does not match the digitmap.
Test Failed: The calling party does not dial a number	The calling party emulation port does not automatically emulate number dialing after detecting the dial tone. Possible cause: The internal processing mechanism of the system encounters an error.
Test Failed: Busy tone is played after the calling party dials a number	The calling party emulation port detects the busy tone after number dialing. Possible causes: The dialed number is busy or the dialed number is incorrect.
Test Failed: The calling party does not communicate with the called party after	The call is not set up. Possible cause: The called party does not pick up the phone.

Test Result	Description
dialing a number	
Test Failed: Release before pick-up of the calling party	The call is released before the calling party and called party enter a conversation. Possible cause: The signaling processing mechanism encounters an error.
Test Failed: The number matching of the calling party is not complete	The calling party emulation port fails to match the DTMF number sent by the called party after entering the conversation. Possible causes: The called party does not press the DTMF number or the DTMF number is lost during the transmission.
Test Failed: The number sending of the calling party is not complete	The called party does not complete the sending of the DTMF number to the calling party emulation port after entering the conversation. Possible cause: The called party hangs up the phone before the sending of the DTMF number is completed.
Test Failed: The number matching of the calling party fails	The DTMF number sent from the called party is incorrect.
Test Failed: The calling port is abnormal	The calling party emulation port is faulty. Possible causes are as follows: • The board is faulty. • The board is removed. • The calling party emulation port is faulty.

Table 1-18 Results of a called party emulation test

Test Result	Description
Test Failed: The phone of the called party does not ring	The called party emulation port does not receive any call. Possible causes: The calling party dials the wrong number or the signaling transmission encounters an error.
Test Failed: The called party does not pick up the phone off the hook	The called party emulation port does not automatically emulate off-hook after detecting the ringing current. Possible cause: The internal processing mechanism of the system encounters an error.
Test Failed: Busy tone is played after the called party picks up the phone off the hook	The called party emulation port detects the busy tone after off-hook. Possible cause: The calling party hangs up the phone.
Test Failed: The called party does not communicate with the calling party	The called party emulation port cannot enter the conversation after off-hook. Possible cause: The called party emulation port

Test Result	Description
	emulates off-hook so slowly that the calling party has hung up the phone.
Test Failed: Release before pick-up of the called party	The call is released before the calling party and called party enter a conversation. Possible cause: The signaling processing mechanism encounters an error.
Test Failed: The number matching of the called party is not complete	The called party emulation port fails to match the DTMF number sent by the calling party after entering the conversation. Possible causes: The calling party does not press the DTMF number or the DTMF number is lost during the transmission.
Test Failed: The number sending of the called party is not complete	The calling party does not complete the sending of the DTMF number to the called party emulation port after entering the conversation. Possible cause: The calling party hangs up the phone before the sending of DTMF number is completed.
Test Failed: The number matching of the called party fails	The DTMF number sent from the calling party is incorrect.
Test Failed: The called port is abnormal	The called party emulation port is faulty. Possible causes are as follows: • The board is faulty. • The board is removed. • The called party emulation port is faulty.

1.16.1.3 Principle Demonstration of the Call Emulation Test

The principle demonstration for a call emulation test includes three parts: calling party emulation test, called party emulation test and calling and called party emulation test.

NOTE

See the following the principle demonstration. In actual applications, commands are executed to start calling party emulation test, called party emulation test and calling and called party emulation test, in which, port IDs are set according to specific devices.

1.16.1.4 Configuring the Call Emulation Test

Prerequisites

- The voice service must be configured.
- A normal phone must be provided.

Procedure

Step 1 Run the **simulate call parameter** command to configure the parameters for the call emulation test.

The type of the call emulation alert tone and DTMF number must be set before the automatic call emulation test is performed on the port.

By default, the type of the call emulation alert tone is voice announcement.

Step 2 Run the **display simulation call parameter** command to query the parameters configured for the current call emulation test.

Step 3 Start a call emulation test. You can start a calling party emulation test, a called party emulation test, or a calling and called party emulation test based on actual requirements.

- To start a calling party emulation test, run the **simulate call start caller** command.
- To start a called party emulation test, run the **simulate call start callee** command.
- To start a calling and called party emulation test, run the **simulate call start call** command.

----End

Result

After completing the call emulation test, the device directly outputs the test result. The test result is used to check whether the call is normal.

1.16.2 POTS User Loop Line Test

A POTS user loop line test is used to test the electrical indicators of the line from the test device (an access node) to a phone. When users' POTS services are faulty, POTS user loop line tests can be performed to test the performance and electrical indicators of the loop line to diagnose whether the loop line is faulty.

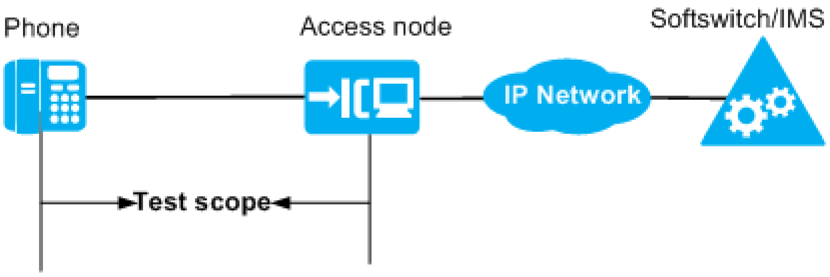
NOTE

The to-be-tested POTS port must not be faulty.

Networking Application

Figure 1-141 shows the example network of a POTS user loop line test. A POTS user loop line test is used to locate faults on the line from an access node to a phone.

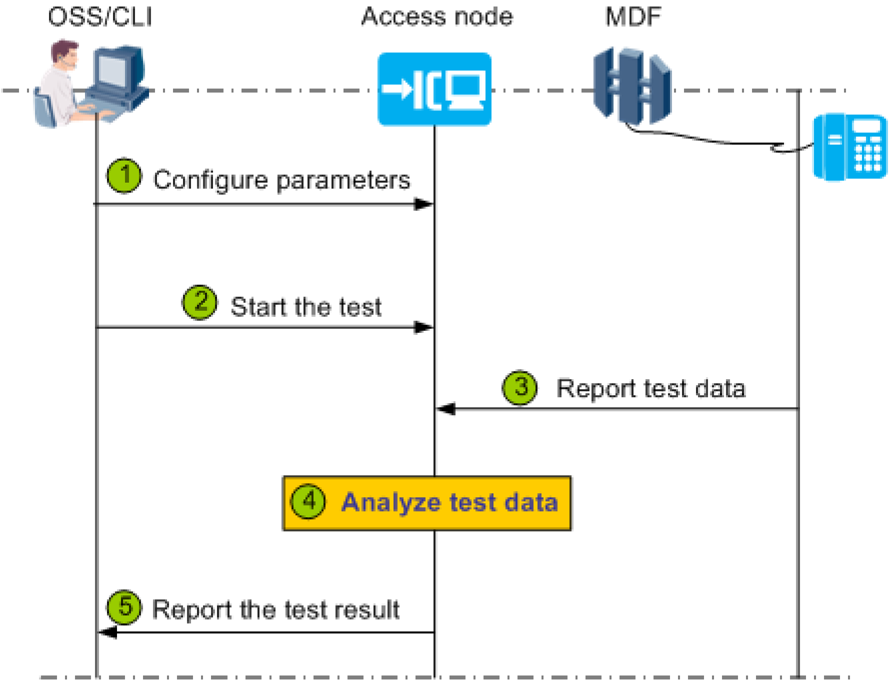
Figure 1-141 Example network of a POTS user loop line test



Test Procedure

Figure 1-142 shows the test procedure of a POTS user loop line test.

Figure 1-142 Test procedure of a POTS user loop line test



1. (Optional) Maintenance engineers set the parameters for the access node on the NMS or remotely log in to the access node from a management PC to set the parameters.

   NOTE

   Generally, the default parameter values are used, and maintenance engineers do not need to set them.

   - In test mode, run the **pots test-para** command to set the physical layer parameters.

     During a loop line test, to avoid affecting the services and functions of the live network, it is necessary to set the physical layer parameters to control the electrical indicators, such as the maximum and minimum voltages supported by the test.

   - In test mode, run the **pots loop-line-threshold** command to set thresholds of the test.

     The access node uses these thresholds as the criteria to check whether the line is faulty when analyzing the test data.

2. Maintenance engineers start a loop line test by using the NMS or running the **pots loop-line-test** command.

   If users (connected to the access node) are making calls during a loop line test, maintenance engineers can cancel, forcibly perform, or delay the test based on actual conditions.

   NOTE

   If maintenance engineers forcibly perform the test, services on the port are interrupted and users' telephone services are affected. Therefore, exercise caution when performing this operation.

3. The access node collects the data of the test items. Table 1-19 lists the loop line test items.

Table 1-19 Loop line test items

Test Item	Specific Test Item

Test Item	Specific Test Item
Voltage	A->G DC voltage
	B->G DC voltage
	A->B DC voltage
	A->G AC voltage
	B->G AC voltage
	A->B AC voltage
	A->G AC frequency
	B->G AC frequency
	A->B AC frequency
Resistance	A->ground insulation resistance
	B->ground insulation resistance
	A->B insulation resistance (low)
	B->A insulation resistance (low)
	A->B insulation resistance (high)
	B->A insulation resistance (high)
Capacitance	A->ground capacitance
	B->ground capacitance
	A->B capacitance (low)
	A->B capacitance (high)
Conductance	A->ground conductance
	B->ground conductance
	A->B conductance (low)
	A->B conductance (high)
Susceptance	A->ground susceptance
	B->ground susceptance
	A->B susceptance (low)
	A->B susceptance (high)
Current	A->ground DC current
	B->ground DC current
	A->B DC current
	B->A DC current

Test Item	Specific Test Item
	A->ground AC current
	B->ground AC current
	A->B AC current
	B->A AC current

4. The access node analyzes the collected data according to the algorithm, and outputs the test conclusion.
5. The access node reports the test conclusion, and maintenance engineers diagnose whether the tested line is faulty based on the test conclusion.

Test Conclusion

Table 1-20lists the loop line test conclusions.

Table 1-20 OLT loop line test conclusions

Item	Conclusion
Line state	Normal
	A->ground AC voltage is hazardous to persons
	B->ground AC voltage is hazardous to persons
	AB->ground AC voltage is hazardous to persons
	A->ground EMF AC voltage exist
	B->ground EMF AC voltage exist
	AB->ground EMF AC voltage exist
	A->ground abnormal AC voltage exist
	B->ground abnormal AC voltage exist
	AB->ground abnormal AC voltage exist
	A->ground DC voltage is hazardous to persons
	B->ground DC voltage is hazardous to persons
	AB->ground DC voltage is hazardous to persons
	AC voltage between A line and B line is hazardous to persons
	DC voltage between A line and B line is hazardous to persons
	A->ground EMF DC voltage exist
	B->ground EMF DC voltage exist
	AB->ground EMF DC voltage exist

Item	Conclusion
	A->ground abnormal DC voltage exist
	B->ground abnormal DC voltage exist
	AB->ground abnormal DC voltage exist
	A line grounding
	B line grounding
	AB line grounding
	A->ground resistance fault
	B->ground resistance fault
	AB->ground resistance fault
	A->ground resistance leak
	B->ground resistance leak
	AB->ground resistance leak
	AB->ground poor insulation
	AB->ground capacitance leak
	A->ground capacitance leak
	B->ground capacitance leak
	Double line break or no terminal
	A line break
	B line break
	Cut off in MDF (that is, a line cut occurs between the main distribution frame and the device)
	Cut off out MDF (that is, a line cut occurs between the main distribution frame and the user side)
	Unknown
PPA test result **NOTE** A passive test termination (PPA) is similar to a test reference point. It is used to detect whether a fault occurs on the loop line between a point and the PPA so that maintenance engineers can locate faults section by section.	PPA not detected
	A->B PPA detected
	B->A PPA detected
	A->B 2 PPA detected
	B->A 2 PPA detected

Item	Conclusion
Terminal status	Off hook
	ETSI Signature or Elec ring circuit
	A-B short
	R-C network (on hook or modem exist)
	Other terminal
	Self mixed in MDF (shorted wires within the same twisted pair, occurring between the main distribution frame and the device)
	Self mixed out MDF (shorted wires within the same twisted pair, occurring between the main distribution frame and the user side)

References

Reference standard and protocol: ITU-T G.996.2 Single-ended line testing for digital subscriber lines (DSL)

1.16.3 POTS User Circuit Test

A POTS user circuit test is used to check whether the chip of a POTS board functions normally. If the POTS services are faulty and the loop line works normally, POTS user circuit tests can be used to test the functions (such as the ringing and power feeding) and some parameters (such as the feeding voltage and ringing voltage) of the board circuit to check whether the circuit works normally.

Feature Dependency and Limitation

- The to-be-tested POTS port must not be faulty.
- Only one circuit test can be started on a POTS board at a time.

Test Procedure

1. Maintenance engineers start a circuit test by using the NMS or running the **pots circuit-test** command.

   If users are making calls during a circuit test, maintenance engineers can cancel, forcibly perform, or delay the test based on actual conditions. If maintenance engineers forcibly perform the test, services on the port are interrupted and users' telephone services are affected. Therefore, exercise caution when performing this operation.

   NOTE

   If maintenance engineers forcibly perform the test, services on the port are interrupted and users' telephone services are affected. Therefore, exercise caution when performing this operation.

2. Maintenance engineers check whether the circuit is faulty based on the test results. The results of a circuit test include: Normal, Abnormal, and Not supported. Table 1-21 lists the specific test items and exception description of a circuit test.

Table 1-21 Test items and exception description of a circuit test

Test Item	Exception Description
Digital voltage	Indicates the digital voltage. Users cannot make calls if this item is abnormal.
Low power supply voltage (negative)	Indicates the low-voltage power supply (negative) to the POTS chip. The power consumption of the board increases if this item is abnormal.
High power supply voltage (negative)	Indicates the high-voltage power supply (negative) to the POTS chip. Users cannot make calls or the ringing tone is irregular if this item is abnormal.
Positive power supply voltage	Indicates the high-voltage power supply (positive) to the POTS chip. The ringing tone is irregular if this item is abnormal.
Off hook detective	Indicates the off-hook detection function of the POTS chip. Users cannot make calls if this item is abnormal.
On hook detective	Indicates the on-hook detection function of the POTS chip. Users cannot make calls if this item is abnormal.
A->B feeder voltage	Indicates the output voltage of the POTS chip. The call quality may be impaired if this item is abnormal.
A->ground feeder voltage	
B->ground feeder voltage	
A->B antipole voltage	
Ringing current voltage	Indicates the output ringing voltage of the POTS chip. The ringing tone is excessively low if this item is abnormal.
Ringing current frequency	Indicates the output ringing frequency of the POTS chip. The ringing tone is irregular if this item is abnormal.
Vertical current between A and B	Indicates the output voltage of the POTS chip. The call quality may be impaired if this item is abnormal.
SLIC temperature	Indicates the temperature of the SLIC. The SLIC will be locked and the call will be interrupted if the SLIC temperature is abnormal.
Stop ringing	Indicates the ringing stopping frequency of the POTS chip. Users can hear the ringing tone, but cannot communicate with the peer party after picking up the phone if this item is abnormal.

Test Item	Exception Description
Loop current	Indicates the output current of the POTS chip. The call quality may be impaired if the output current is lower than 18 mA.

1.16.4 POTS Port Loop Test

A POTS port loop test is used to test the hardware and configurations related to POTS services during device installation or before POTS service provisioning. It helps reduce the number of site visits and minimize maintenance costs.

Overview

Maintenance engineers can locally start a POTS port loop test on the device, or remotely log in to the device and then start the test. A POTS port loop test consists of two parts:

- Device hardware test: This test targets at access nodes that are not yet provisioned with the voice service. When an access node is newly deployed, the hardware of the voice module needs to be tested to evaluate the hardware capabilities in supporting future voice services.
- Device service test: This test targets at access nodes that are already provisioned with the voice service. Before voice services are provisioned from the access node to a user, a device service test is performed to determine whether the voice service capabilities are supported by the access node.

NOTE

- Do not pick up the phone during a loop test. Otherwise, the test results will be incorrect.
- If the dialup mode of a PSTN port is set to **DTMF-only**, no loop test can be started on the PSTN port.

Device Hardware Test

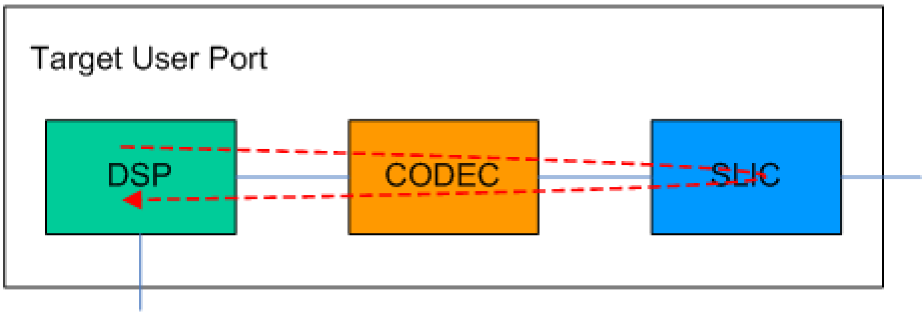
The device hardware test involves the following items:

- Off-hook detection
- On-hook detection
- Ringing and ringing stopping detection on a service port
- Speech path detection

For the first 3 test items, a POTS board emulates off-hook, on-hook, ringing, and ringing stopping, while the control board of the device performs the loop test on the POTS board. The last test item (speech path detection) is used to verify the service processing capability of the chip through service loopbacks. A speech path detection involves 3 loopback tests: SLIC loopback test, CODEC loopback test, and DSP loopback test.

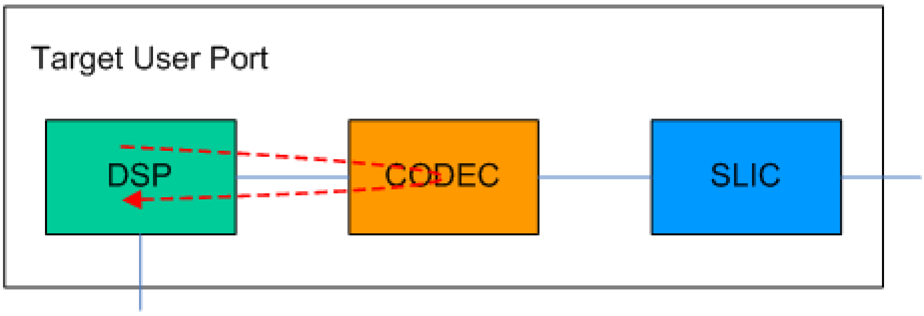
- In a SLIC loopback test, a POTS port is connected to a DSP channel, and the DSP chip generates the DTMF authentication code, which is sent to the TDM side. The DSP chip then detects the DTMF returned from the TDM side to check whether the channel between the DSP, CODEC, and SLIC is normal, as shown in Figure 1-143.

Figure 1-143 Working principles of a SLIC loopback test



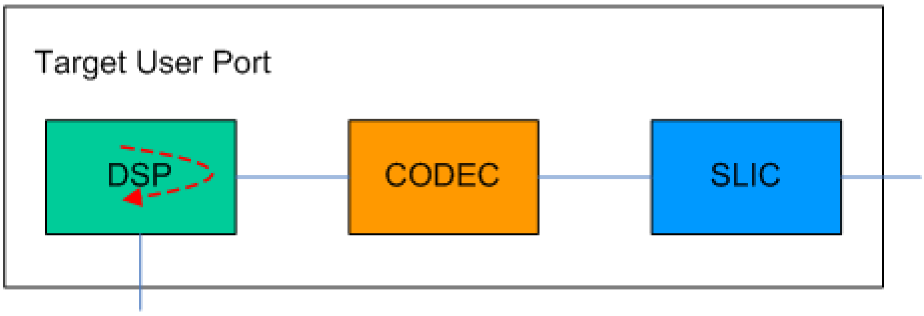
- In a CODEC loopback test, the CODEC loopback is set to the network side (remote loopback), and the DSP chip generates the DTMF authentication code, which is sent to the TDM side. The DSP chip then detects the DTMF returned from the TDM side to check whether the channel between the DSP and the CODEC is normal, as shown in Figure 1-144.

Figure 1-144 Working principles of a CODEC loopback test



- In a DSP loopback test on the TDM side, a DSP channel is looped back from the TDM side to the IP side, and the DSP chip generates the DTMF authentication code, which is sent to the TDM side. The DSP chip then detects the DTMF returned from the TDM side to verify the transmit and receive functions of the TDM side, as shown in Figure 1-145.

Figure 1-145 Working principles of a DSP loopback test on the TDM side



NOTE

In test mode, run the **pots path-test** *frameid/slotid* **port** *portid* **type hardware** command to start a device hardware test. The test results will be displayed after the test is completed. Engineers diagnose whether the system functions normally based on the test results.

Device Service Test

The principles of a device service test are similar to those of a call emulation test. In a device service test, the device acts as both the calling party and called party (that is, the dest port on the device calls another assistant port on the same device), and the system checks whether the configurations are correct. User ports do not enter the conversation state during a POTS port loop test. When the called port detects the ringing, the calling port emulates user on-hook, and therefore, no charge record is generated. A device service test checks:

- Device interface data, including the signaling data (such as MG interface data, protocol parameters, and call server parameters)
- User port data, including the media gateway (MG) IDs and terminal IDs (TIDs) of H.248 users and user accounts and service rights of SIP users

NOTE

In test mode, run the **pots path-test** *frameid/slotid* **port** *portid* **type business** command to start a device service test. The test results will be displayed after the test is completed. Engineers diagnose whether the system functions normally based on the test results.

1.16.5 Line Matching Test

A line matching test is used to check wiring connectivity between the subscriber line and the main distribution frame (MDF), and check wiring sequence.

NOTE

- A line matching test may cause service interruption. Therefore, exercise caution when starting this test.
- During a line matching test, do not perform any service operations on the board to be tested. If a service operation is needed for such a board, reset the board after the test completes so that services can be restored.

Value

Previously, a line matching test was implemented for a single line under the cooperation between 2 deployment or maintenance engineers using a multimeter. This process is of low efficiency and high labor costs. Furthermore, subscriber lines are removed and inserted multiple times during the test, which harms the board and line connectors.

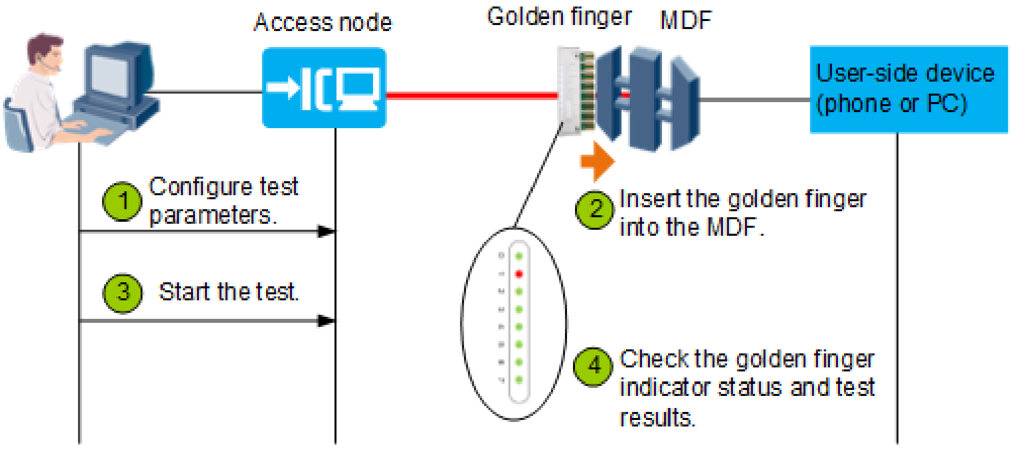
Now, a line matching test can be implemented by using the test tool CLT-Match and golden finger indicator board by only one engineer. Also, lines can be tested by groups. This process dramatically improves test efficiency and correctness. However, damages caused by subscriber line removal and insertion are unavoidable.

In an actual test, the software can substitute for the CLT-Match. Specifically, the software triggers a line matching test. Then, using together with the golden finger indicator board, subscriber lines do not need to be removed or inserted. This process improves test efficiency and protects the board and subscriber lines.

Procedure

As shown in Figure 1-146, the red line indicates the route from the subscribe line to the MDF, which is tested in a line matching test for its wiring connectivity and sequence.

Figure 1-146 Procedure


Access node
Golden finger
MDF
User-side device (phone or PC)
1 Configure test parameters.
3 Start the test.
2 Insert the golden finger into the MDF.
4 Check the golden finger indicator status and test results.

1. The deployment or maintenance engineer logs in to the device and runs the **line-match-config** command to configure parameters for a line matching test. In the configuration, the number of line matching test cycles, the blinking rate of the line matching test indicator, and the line matching test mode can be specified.
2. The deployment or maintenance engineer inserts the golden finger indicator board into the equipment-side MDF module of the corresponding wiring module.
3. The deployment or maintenance engineer runs the **line-match-test** command to start a line matching test. Only one board can be tested at a time.
4. The deployment or maintenance engineer observes the indicator blinking status on the MDF side to determine whether wiring is correct.
   - If the indicator blinks green, the line connectivity is normal.
   - If the indicator blinks red, the line connectivity is abnormal and a manual check is required.
   - If the one-by-one test mode is set and indicators blink in the sequence of ports, wiring sequence is correct.
   - If the one-by-one test mode is set and indicators do not blink in the sequence of ports, wiring sequence is incorrect. This may be caused by shorted wires and a manual check is required.

1.16.6 Search Tone Test

A search tone test is a simple line fault locating function intended for maintenance engineers. In a search tone test, the test module sends voice signals with the specific frequency and amplitude to a line, and then maintenance engineers use a receiver or a dedicated device to detect the signals on the line. In addition, search tone tests can help maintenance engineers pinpoint the specific line among multiple user lines.

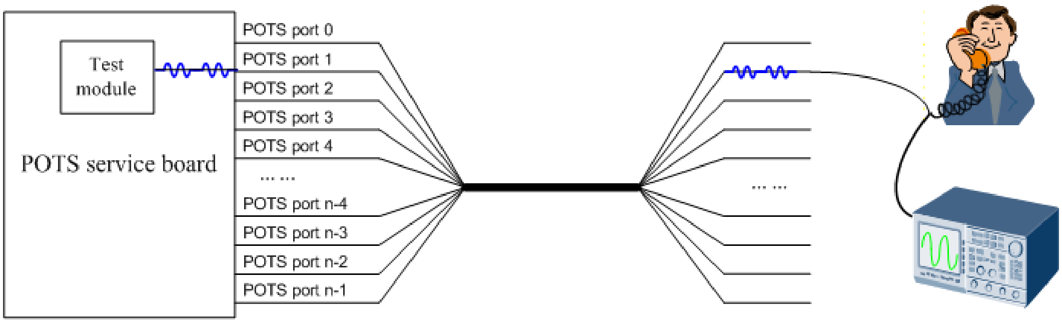
NOTE

- A search tone test can be performed even when a port is powered off.
- A search tone test cannot be performed when another test task (such as a POTS user circuit test, POTS user loop line test, call emulation test, signal tone test, or POTS port loop test) is in progress.
- A search tone test cannot be performed when a port is in the prohibit state.

Test Procedure

Figure 1-147 shows the principles of a search tone test.

Figure 1-147 Principles of a search tone test



1. The test module sends voice signals with the specific frequency and amplitude (as underlined by blue wave lines in Figure 1-147) to a line after a search tone test is started.

   In test mode, run the **pots search-tone-test** *frameid/slotid/portid* **test-flag enable** command to start a search tone test. If users are making calls during a search tone test, maintenance engineers can cancel or forcibly perform the test based on actual conditions.

NOTE

If maintenance engineers forcibly perform the test, services on the port are interrupted and users' telephone services are affected. Therefore, exercise caution when performing this operation.

2. Maintenance engineers use a receiver or a dedicated device connected to the other end of the line to detect whether the voice signals can be received. If the voice signals can be received, the line works normally.
3. (Optional) Stop a search tone test.

   Run the **pots search-tone-test** *frameid/slotid/portid* **test-flag disable** command to stop a search tone test.

NOTE

In a search tone test, the duration of playing the search tone needs to be set based on actual requirements.

- If the preset duration does not expire, maintenance engineers can run this command to manually stop the test.
- If the preset duration has expired, the system automatically stops the test.

1.16.7 Signal Tone Test

In a signal tone test, the system sends the signal tone signals to a specific port of a POTS board and makes the port loop back the signals, and then checks whether the loopback signals can be detected. This test function helps maintenance engineers check whether the system can normally process the detection of the user off-hook and signal tone and locate hardware faults related to the user off-hook and signal tone playing.

A signal tone test includes the following types, as listed in the following table.

Test Type	Requiring Coordination of Upper-Layer Device?
Busy tone test	The coordination of the upper-layer device

Test Type	Requiring Coordination of Upper-Layer Device?
Ringback tone test	is not required.
Dial tone test	The dial tone test and special dial tone test include the following modes. • **out-of-service**: In this mode, the test is performed on the device, which is independent of the upper-layer device. If the device is not connected to the upper-layer device, use this mode. • **in-service**: In this mode, the device coordinates with the upper-layer device to perform the test. The upper-layer device controls the playing of the dial tone or special dial tone.
Special dial tone test (that is, the dial tone test started by the device based on different service requirements, such as call forwarding-unconditional)	

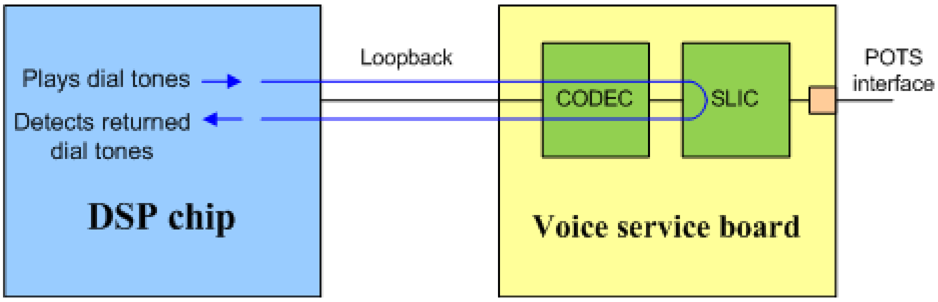
NOTE

- The POTS port supports local loopbacks.
- The POTS port must be in the on-hook state during a signal tone test.
- A signal tone test cannot be performed if the port is in the **prohibit** or **loopback** state.
- A signal tone test cannot be performed when another test task (such as a POTS user circuit test, POTS user loop line test, call emulation test, search tone test, or POTS port loop test) is in progress.

Test Procedure

Figure 1-148 shows the test procedure of a signal tone test. As shown in Figure 1-148, the subscriber line interface circuit (SLIC) supplies feeding current to the telephone, sends voice frequency, generates ringing, detects off-hook signals and on-hook signals, and processes analog signals. The CODEC performs the conversion between analog signals and digital signals.

Figure 1-148 Test procedure of a signal tone test



1. Maintenance engineers start a signal tone test on a POTS user port by using the CLI or NMS.
   - In test mode, run the **pots signal-tone-test** *frameid/slotid/portid* **signal-tone busy-tone** command to start a busy tone test.

   - In test mode, run the **pots signal-tone-test** *frameid/slotid/portid* **signal-tone ringback-tone**command to start a ringback tone test.
   - In test mode, run the **pots signal-tone-test** *frameid/slotid/portid* **signal-tone dial-tone**command to start a dial tone test.
   - In test mode, run the **pots signal-tone-test** *frameid/slotid/portid* **signal-tone special-dial-tone** command to start a special dial tone test.

2. The DSP chip plays the signal tone.
   - For the busy tone test or ringback tone test, the device requests the DSP chip to issue the busy tone or ringback tone to the POTS port after the test is started.
   - For the dial tone test or special dial tone test:
     - If the test is started in **out-of-service** mode, the POTS port emulates the user off-hook. After the device detects the off-hook signal, it requests the DSP chip to issue the dial tone or special dial tone to the POTS port.
     - If the test is started in **in-service** mode, the POTS port emulates the user off-hook. After the device detects the off-hook signal, it reports the signal to the softswitch/IMS, and the softswitch/IMS requests the DSP chip to issue the dial tone or special dial tone to the POTS port.

NOTE

You can start the test in **in-service** mode only after you have configured the users.

3. The POTS board loops back the signal tone signals (as shown by the blue lines in Figure 1-148).
4. If the DSP chip can detect the loopback signals, the system runs normally.

   If any exception occurs during the test, the following methods can be used for troubleshooting.

Exception	Troubleshooting Method
No looped back signal is detected.	• Run the **display pstn state** command to query the port status to check whether the port is faulty. • Run the **display pstn state** command to query the port status to check whether the port is busy. • Run the **display dsp state** command to query the DSP channel status to check whether the DSP resources are sufficient.
The delay of the signal tone is too long.	• Run the **display cpu** command to query the CPU usage to check whether the system is overloaded. • If the test is started in **in-service** mode, check whether the interaction between the device and the softswitch/IMS is delayed.

5. (Optional) Stop a signal tone test.

Run the **pots signal-tone-test** *frameid/slotid/portid* **test-flag disable** command to stop a signal tone test.

NOTE

In a signal tone test, the duration of playing the signal tone needs to be set based on actual requirements.

- If the preset duration does not expire, maintenance engineers can run this command to manually stop the test.
- If the preset duration has expired, the system automatically stops the test.

1.16.8 RTCP Statistics

Complying with the H.248/SIP protocol, the softswitch/IMS, during and at the end of a call, can query the RTCP statistics of a user, including the number of transmitted RTP packets, bytes of transmitted RTP packets, number of received RTP packets, bytes of received RTP packets, number of lost transmitted packets, number of lost received packets, network jitter, and network loop delay.

The access device reports its real-time statistics to the softswitch/IMS when the softswitch/IMS issues signaling to query the statistics. Then, the softswitch/IMS or the OSS system can manage the quality monitoring based on the statistics.

1.16.9 Signaling Tracing

This topic describes the purpose and implementation principles of signaling tracing fault diagnosis feature.

NOTE

Based on your requirements, VBD fault diagnosis may obtain some contents of users' communications (integrity communication contents are not obtained and user information will not be disclosed) for the purpose of safeguarding network operations and protecting services. Huawei alone is unable to collect or save the content of users' communications. You must comply with the laws and regulations of the countries concerned for using the VBD fault diagnosis feature. You are obligated to take considerable measures to ensure that the content of users' communications is fully protected when the content is being used and saved.

Introduction

After the OLT is running on the network and voice service failures occur, such as poor communication quality, low fax/modem service success rate, one-way audio, and no audio, easy signaling tracing (instead of remote packet capture) can be used to obtain voice signaling streams and further these signaling streams can be used for fast fault locating.

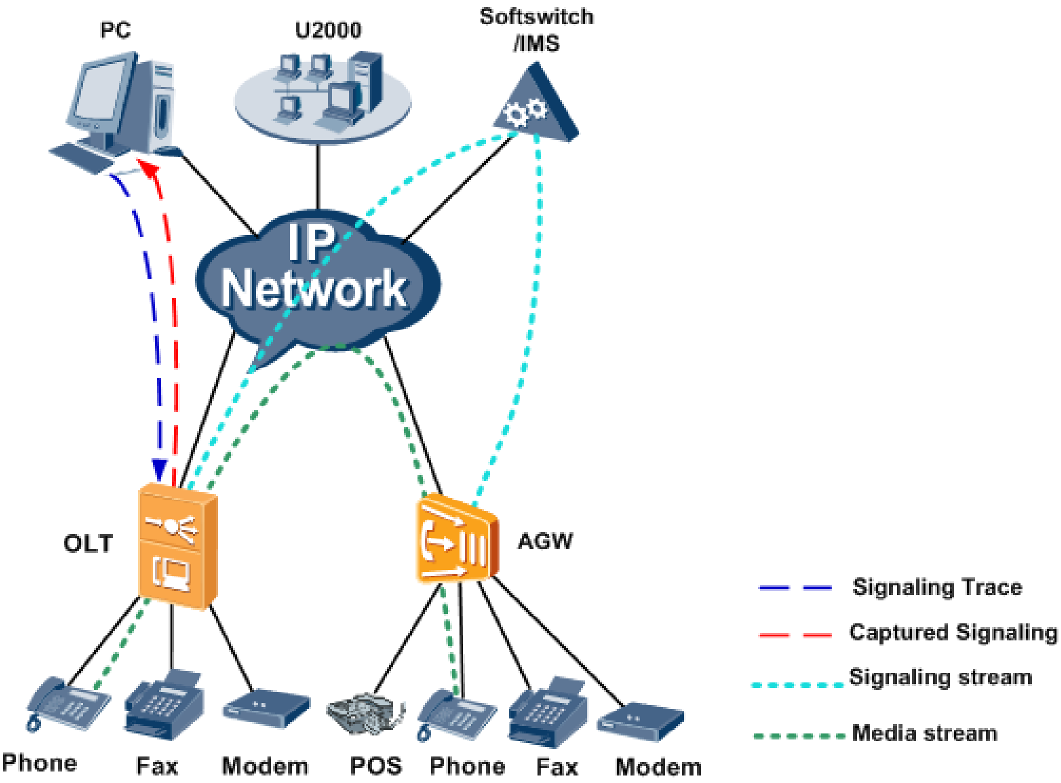
Signaling tracing features:

- Remote fault locating: Fault locating can be remotely implemented, eliminating the need of site visits.
- Fast fault locating: Remote management or packet capture software is not needed, shortening fault locating duration.
- Accurate fault locating: Voice signaling tracing can be performed based on fault symptoms.
- Intelligent fault locating: Intelligently associates with the tracked tasks, which lowers skills required for maintenance personnel and simplifies operations.
- High security and reliability: The obtained signaling file does not contain any user privacy, such as the dialed telephone number, name, SMS, and FSK.

Networking Applications

Figure 1-149 shows the networking of signaling tracing.

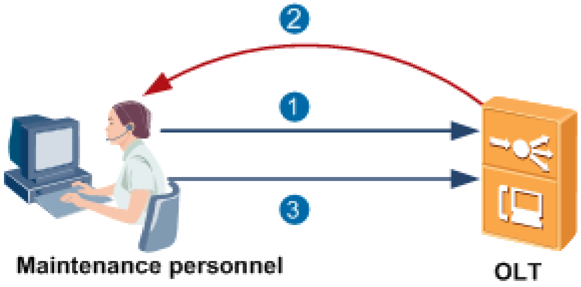
Figure 1-149 Networking of signaling tracing



Signaling tracing takes effect on signaling exchanged between the IMS and OLT. The O&M engineer obtains the traced signaling by telnetting to the OLT from the PC.

Figure 1-150 shows the process of obtaining voice signaling.

Figure 1-150 Process of obtaining voice signaling

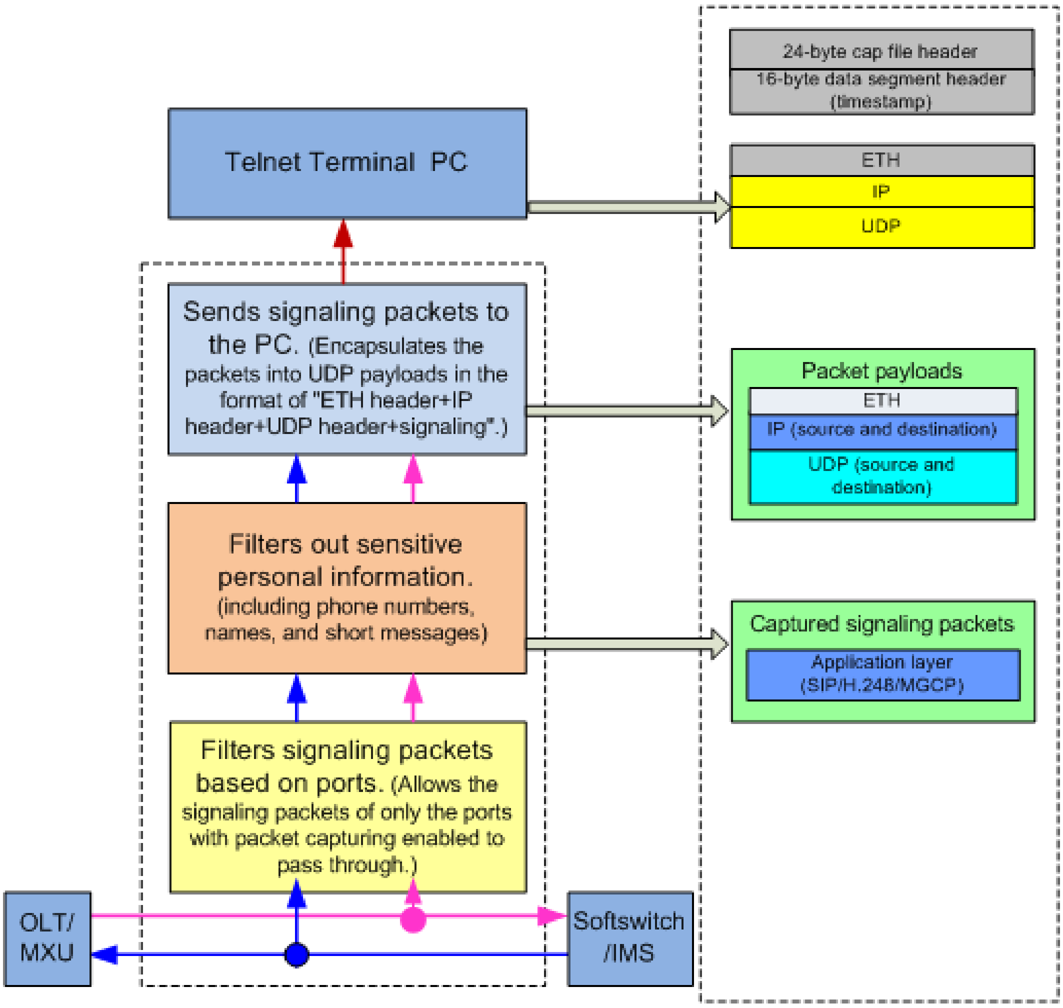


1. The O&M engineer telnets to the OLT from the PC and starts voice signaling tracing through the CLI.
2. The OLT sends the obtained voice signaling to the specified PC.
3. The O&M engineer stops obtaining voice signaling.

Working Principles

Figure 1-151 shows the application of authorized signaling tracing.

Figure 1-151 Application of authorized signaling tracing


24-byte cap file header
16-byte data segment header (timestamp)
Telnet Terminal PC
ETH
IP
UDP
Sends signaling packets to the PC. (Encapsulates the packets into UDP payloads in the format of "ETH header+IP header+UDP header+signaling".)
Packet payloads
ETH
IP (source and destination)
UDP (source and destination)
Filters out sensitive personal information. (including phone numbers, names, and short messages)
Captured signaling packets
Application layer (SIP/H.248/MGCP)
Filters signaling packets based on ports. (Allows the signaling packets of only the ports with packet capturing enabled to pass through.)
OLT/ MXU
Softswitch /IMS

After authorized signaling tracing is enabled:

1. The OLT filters packets based on ports. The OLT captures the signaling packets only on the ports with signaling packet capturing enabled.
2. The OLT filters out sensitive personal information, including phone numbers, names, short messages, FSK data, and DTMF data, from the captured signaling packets.
3. The OLT encapsulates the signaling packets into UDP payloads in the format of "ETH header+IP header+UDP header+signaling" and sends the UDP payloads to the PC.
4. The O&M engineer analyzes the obtained voice signaling and locates faults on the telnet terminal.

Procedure

Perform the following to obtain the voice signaling to locate a voice service fault quickly.

1. Run the **diagnose** command to enter diagnose mode from privilege mode or global config mode.
2. Run the **signal trace** command to start a signaling tracing task.
3. Run the **undo signal trace** command stop signaling tracing manually or the system stops signaling tracing automatically (after the preset signaling tracing duration expires).

Result

After signaling tracing, run the **display signal trace** command to query the signaling tracing information about the current port.

1.16.10 VBD Fault Diagnosis

This topic describes the purpose and implementation principles of the voice band data (VBD) fault diagnosis feature.

NOTE

Based on your requirements, VBD fault diagnosis may obtain some contents of users' communications (the information cannot restore the integrity communications contents) for the purpose of safeguarding network operations and protecting services. Huawei alone is unable to collect or save the content of users' communications. You must comply with the laws and regulations of the countries concerned for using the VBD fault diagnosis feature. You are obligated to take considerable measures to ensure that the content of users' communications is fully protected when the content is being used and saved.

Introduction

The VBD fault diagnosis feature enables the OLT to obtain voice packets, signaling packets, and voice control packets when a narrowband service fault occurs on the OLT. Then, O&M engineers can use these packets for rapid fault location. The narrowband service fault can be:

- Poor quality of voice communication
- High failure ratio of the fax or modem service
- One-way audio or no audio

NOTE

Voice control packets contain higher-layer communication protocols used between the control board and service boards of the OLT as well as digital signal processor (DSP) commands for processing internal board data.

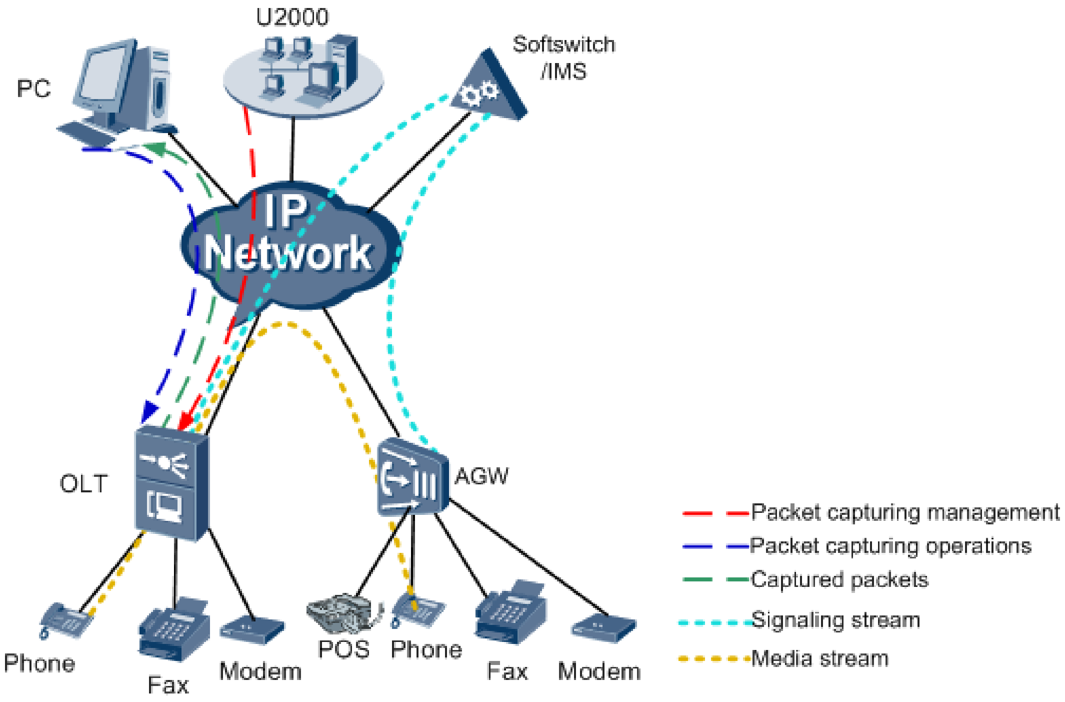
The VBD fault diagnosis feature supports:

- Remote fault location: removes the need of onsite fault location.
- Rapid fault location: enables O&M personnel to obtain voice packets, signaling packets, and voice control packets based on fault symptoms, thereby improving fault location efficiency.
- Intelligent fault locating: Intelligently associates with the packet capturing tasks, which lowers skills required for maintenance personnel and simplifies operations.
- Secure and reliable data obtaining: The obtained voice packets, signaling packets, and voice control packets do not contain any user privacy information, such as called numbers, names, short messages, or frequency shift keying (FSK) data.

Networking Applications

Figure 1-152 shows the networking of the voice VBD fault diagnosis feature.

Figure 1-152 Networking of the VBD fault diagnosis feature



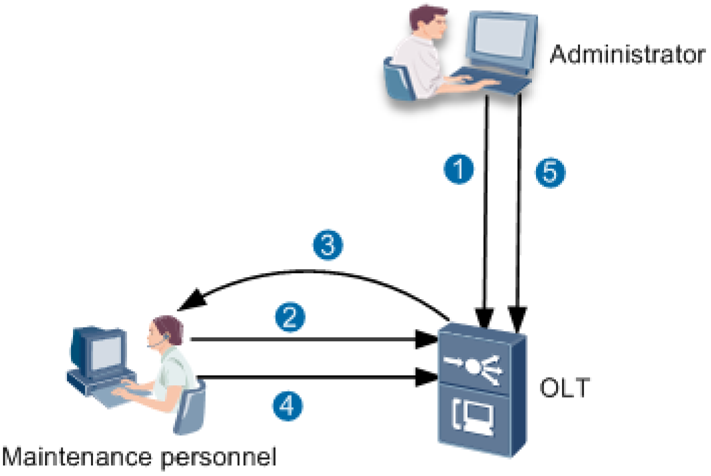
Carrier's administrator grants packet capturing rights to the O&M personnel. Then, the O&M personnel capture packets on a specified PC (packet capturing server). The OLT filters out users' communication content from the captured packets, encapsulates the filtered packets into UDP packet payloads, and sends the UDP packets to the PC.

NOTE

Authorized packet capturing complies with local laws and regulations and is performed carrier's authorization. Authorized packet capturing is used to capture voice packets, signaling packets, and voice control packets that do no contain user sensitive information and the captured data is only used for fault location.

Figure 1-153 shows the process of obtaining voice packets.

Figure 1-153 Process of obtaining voice packets



1. The carrier's administrator uses the administrator account to remotely log in to the OLT from the PC and grants packet capturing rights to the O&M personnel.
2. The O&M personnel enable a general voice packet capturing tool, such as wireshark, on the PC, use a non-administrator account to remotely log in to the OLT, capture voice packets through the maintenance port.

NOTE

To safeguard network operations and protect services, the O&M personnel can use only a non-administrator account to log in to the OLT.

3. The OLT sends the obtained voice packets to the PC.
4. The O&M personnel stop capturing voice packets.
5. The carrier's administrator cancels the packet capturing permission.

Basic Concepts Related to Authorized Media Stream Packet Capturing

Capturing all voice packets of a call easily discloses user privacy information. To protect user privacy, the OLT provides differentiated VBD fault diagnosis modes. By doing so, VBD fault diagnosis can be enabled without bringing the risk of disclosing user privacy information.

Table 1-22 lists the VBD fault diagnosis modes that the OLT provides.

Table 1-22 VBD fault diagnosis modes

Mode	Description	Usage Scenario
Full packet capturing	The OLT does not filter or replace any information in the captured Real-Time Transport Protocol (RTP) or time division multiplexing (TDM) tracing packets.	This mode applies when the captured packets do not contain the content of users' communications. In this mode, packets can be captured as many as possible, which facilitates fault location. This mode is used to diagnose fax or modem negotiation failures.
Fuzzy packet capturing	The OLT retains the minimum data that identifies the fax or modem signal tone in RTP and	This mode applies when the OLT cannot determine whether a user will initiate a VBD call, or when the content

Mode	Description	Usage Scenario
	TDM tracing packets, which prevents the original content of users' communications from being restored. Specifically, the OLT retains only 10 ms of data (data generated in a 10-millisecond duration) from every 80 ms of data and erases the 70-ms data. In this way, packets are captured in a discontinuous way and the original content of users' communications cannot be restored.	of users' communications will be transmitted after a VBD call is set up. This mode is used when the local or peer end cannot identify a fax or modem signal tone or a fax or modem negotiation between the local and peer ends fails.
Packet header-only capturing	The OLT captures only the IP, UDP, and RTP headers of RTP packets. It replaces the payloads of the RTP packets with fixed data. In addition, the OLT does not capture TDM tracing packets.	This mode applies when the VBD negotiation process ends. This ensures that no content of users' communications will be captured. This mode is used to locate voice quality faults, such as packet loss, jitter, delay, one-way audio, and no audio.

Working Principles of Authorized Media Stream Packet Capturing

Figure 1-154 and Figure 1-155 show the application of the voice packet capturing modes in a fax or modem call and in a common call (Taking the local ringback tone as an example).

Figure 1-154 Application of the voice packet capturing modes in a fax or modem call

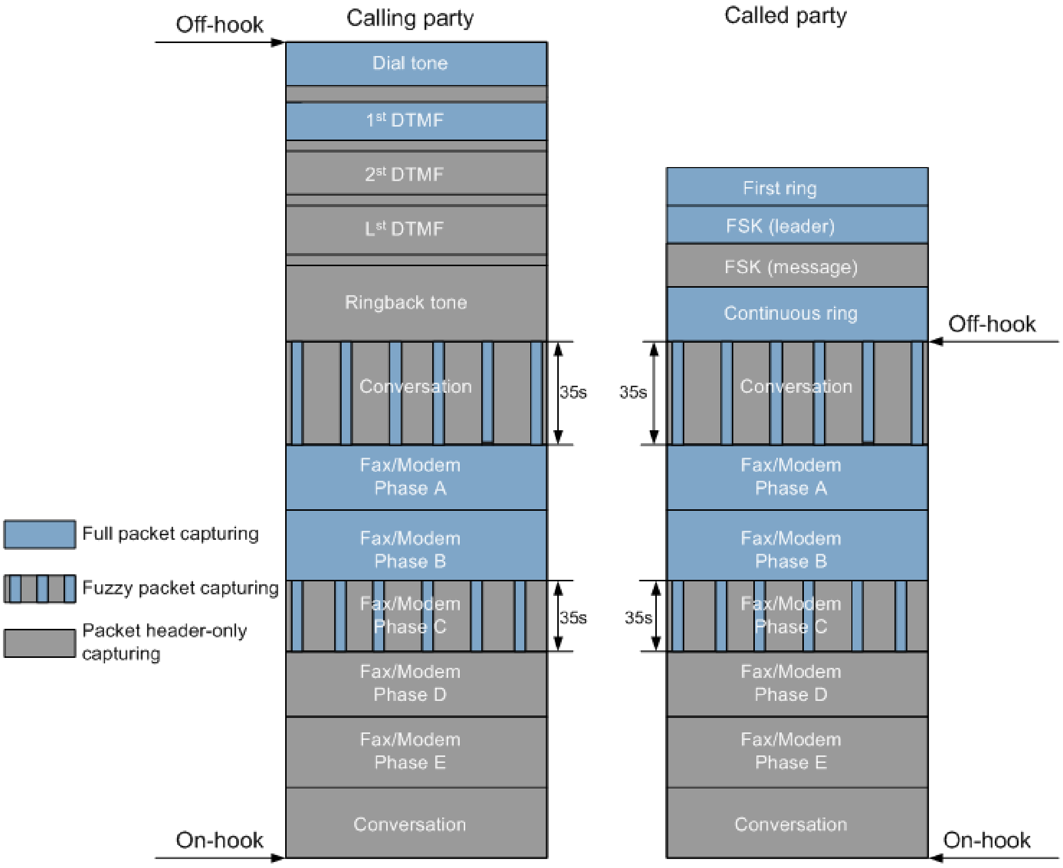
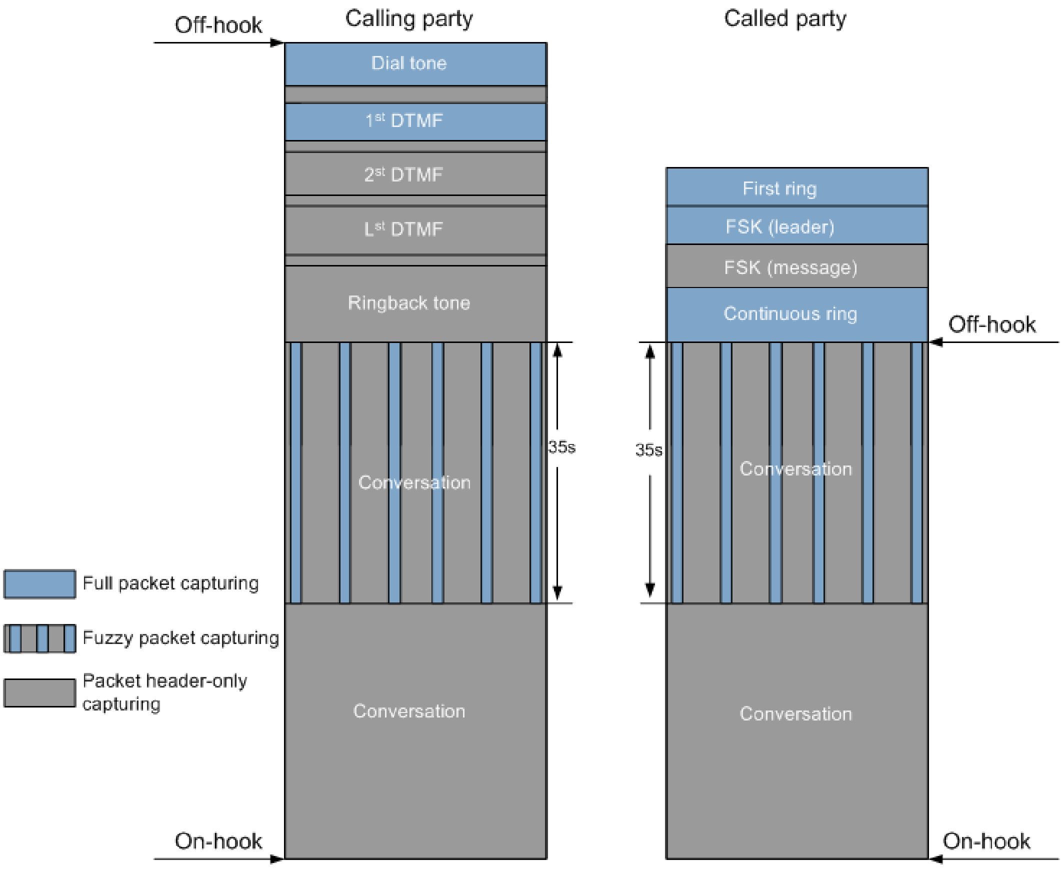


Figure 1-155 Application of the voice packet capturing modes in a common call



For the calling party:

1. After the calling party picks up the phone, the OLT starts the full packet capturing mode.
2. After detecting the first dual tone multiple frequency (DTMF) number, the OLT switches to the packet header-only capturing mode, which prevents disclosure of the user DTMF numbers.
3. When the calling party hears ringback tones, the capturing mode does not changed.
4. After the call is set up between the calling and called parties, the OLT switches to the fuzzy packet capturing mode and starts a fuzzy packet capturing protection timer (35s). The fuzzy packet capturing protection timer limits the fuzzy packet capturing duration and protects user communication security.

NOTE

The call may be a VBD call, during which the VBD signal tone may not be identified. Therefore, the fuzzy packet capturing mode is required to locate faults where the VBD signal tone fails to be identified.

5. If the OLT detects the fax or modem signal tone and the two ends of the fax or modem service are at the negotiation phase, the OLT switches to the full capturing mode because no communication data is involved at the negotiation phase. In this way, the OLT can capture most fault-related packets, which facilitates rapid fault location.

NOTE

This step is involved only in a fax or modem call.

6. Before the negotiation phase ends, the OLT switches to the fuzzy packet capturing mode.

NOTE

This step is involved only in a fax or modem call.

7. When the fuzzy packet capturing protection timer times out, the OLT switches to the packet header-only capturing mode, which protects user communication security.

For the called party:

1. When the phone of the called party rings, the OLT starts the full packet capturing mode.
2. When sending FSK data to the called party, the OLT switches to the packet header-only capturing mode, which prevents the content of users' communications from being captured.
3. After the call is set up between the calling and called parties, the OLT switches to the fuzzy packet capturing mode and starts a fuzzy packet capturing protection timer (35s). The fuzzy packet capturing protection timer limits the fuzzy packet capturing duration and protects user communication security.

NOTE

The call may be a VBD call, during which the VBD signal tone may not be identified. Therefore, the fuzzy packet capturing mode is required to locate faults where the VBD signal tone fails to be identified.

4. If the OLT detects the fax or modem signal tone and the two ends of the fax or modem service are at the negotiation phase, the OLT switches to the full capturing mode because no communication data is involved at the negotiation phase. In this way, the OLT can capture most fault-related packets, which facilitates rapid fault location.

NOTE

This step is involved only in a fax or modem call.

5. Before the negotiation phase ends, the OLT switches to the fuzzy packet capturing mode.

NOTE

This step is involved only in a fax or modem call.

6. When the fuzzy packet capturing protection timer times out, the OLT switches to the packet header-only capturing mode, which protects user communication security.

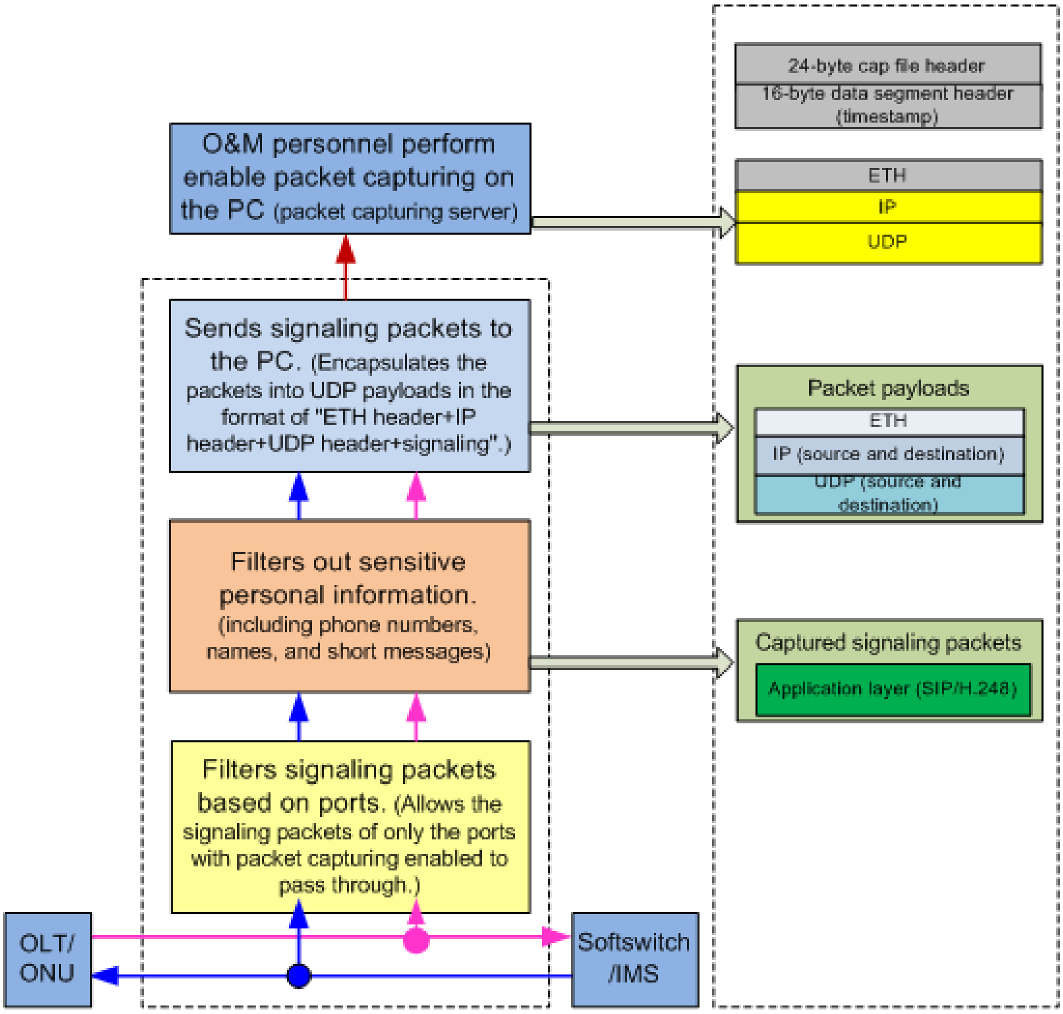
NOTE

When detecting the first DTMF number (or an RFC 2833-compliant DTMF number), the OLT switches to the packet header-only capturing mode to protect the numbers dialed by users.

Working Principles of Authorized Signaling Packet Capturing

Figure 1-156 shows the application of authorized signaling packet capturing.

Figure 1-156 Application of authorized signaling packet capturing


24-byte cap file header
16-byte data segment header (timestamp)
O&M personnel perform enable packet capturing on the PC (packet capturing server)
ETH
IP
UDP
Sends signaling packets to the PC. (Encapsulates the packets into UDP payloads in the format of "ETH header+IP header+UDP header+signaling".)
Packet payloads
ETH
IP (source and destination)
UDP (source and destination)
Filters out sensitive personal information. (including phone numbers, names, and short messages)
Captured signaling packets
Application layer (SIP/H.248)
Filters signaling packets based on ports. (Allows the signaling packets of only the ports with packet capturing enabled to pass through.)
OLT/ ONU
Softswitch /IMS

After authorized signaling packet capturing is enabled, the OLT performs the following operations on the signaling packets, such as SIP, MGCP, or H.248 packets, received from the softswitch/IP multimedia subsystem (IMS) or received by the OLT:

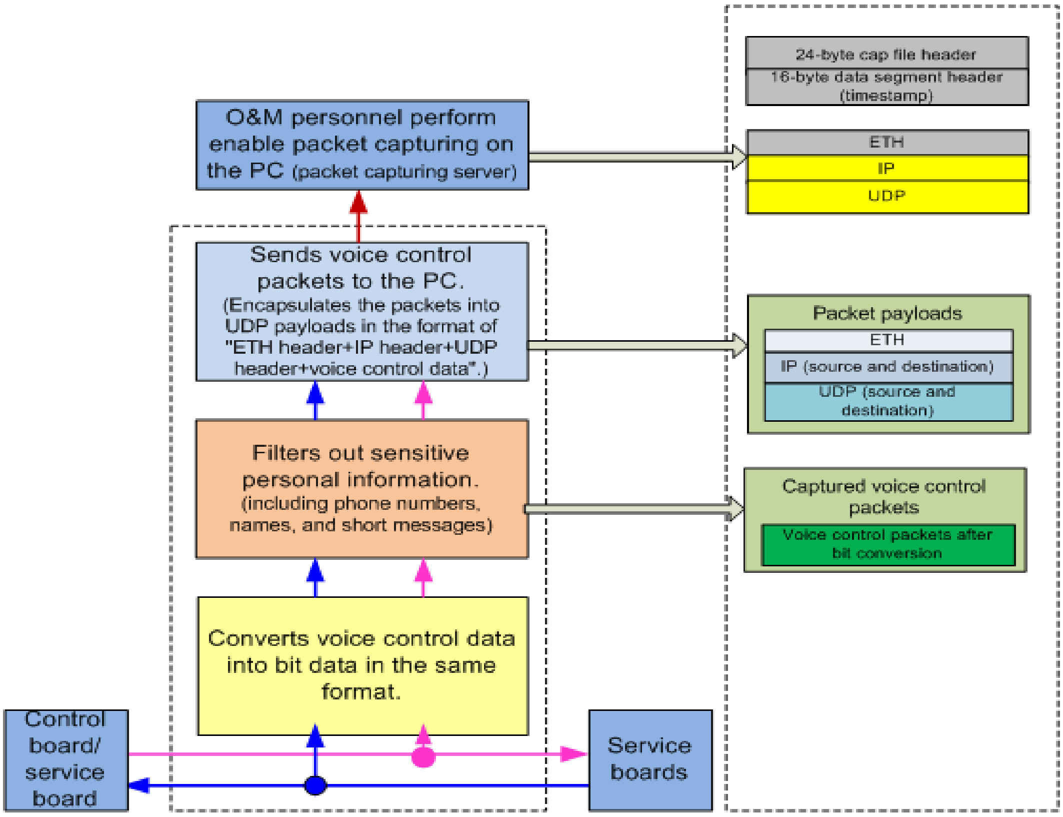
1. Filters packets based on ports. The OLT captures the signaling packets only on the ports with signaling packet capturing enabled.
2. Filters out sensitive personal information, including phone numbers, names, short messages, FSK data, and DTMF data, from the captured signaling packets.
3. Encapsulates the signaling packets into UDP payloads in the format of "ETH header+IP header+UDP header+signaling" and sends the UDP payloads to the PC.

The O&M personnel enable a general voice packet capturing tool, such as wireshark, on the PC to capture voice packets and analyze the voice packets for fault location.

Working Principles of Authorized Voice Control Packet Capturing

Figure 1-157 shows the application of authorized voice control packet capturing.

Figure 1-157 Application of authorized voice control packet capturing


24-byte cap file header
16-byte data segment header (timestamp)
O&M personnel perform enable packet capturing on the PC (packet capturing server)
ETH
IP
UDP
Sends voice control packets to the PC. (Encapsulates the packets into UDP payloads in the format of "ETH header+IP header+UDP header+voice control data".)
Packet payloads
ETH
IP (source and destination)
UDP (source and destination)
Filters out sensitive personal information. (including phone numbers, names, and short messages)
Captured voice control packets
Voice control packets after bit conversion
Converts voice control data into bit data in the same format.
Control board/ service board
Service boards

After authorized voice control packet capturing is enabled, OLT performs the following operations on the packets transmitted between the control board and service boards or between service boards:

1. Converts voice control data into bit data in the same format.
2. Filters out sensitive personal information, including phone numbers, names, short messages, FSK data, and DTMF data, from the captured voice control packets.
3. Encapsulates the voice control packets into UDP payloads in the format of "ETH header+IP header+UDP header+voice control data" and sends the UDP payloads to the PC.

The O&M personnel enable a general voice packet capturing tool, such as wireshark, on the PC to capture voice packets and analyze the voice packets for fault location.

Configuration Procedure

Perform the following steps to capture voice packets:

1. Run the **remote-capture whitelist add** command to add a specified user port on the PC to the remote packet capturing whitelist.

   Only the ports in the remote packet capturing whitelist can be used to capture voice packets.

NOTE

The carrier's administrator must grant the operation rights to the O&M personnel for this step.

2. Run the **remote-capture parameters** command to set the IP address and port number of the PC where voice packets can be captured.
3. Run the **remote-capture trace** command to configure the port or channel for capturing packets and the packet capturing duration, and start voice packet capturing.
4. Run the **undo remote-capture trace** command to manually stop capturing voice packets.
5. After capturing voice packets, run the **remote-capture whitelist delete** command to delete the user port from the remote packet capturing whitelist.

Configuration Results

After the configuration, you can:

- Run the **display remote-capture state** command to query packet capturing status on the port.
- Capture voice packets on the PC where the general voice packet capturing tool, such as wireshark, is enabled.

1.16.11 QoS Alarm

The network quality affects the voice service to a great extent. During a call, the access device monitors the network quality in real time. When the network quality is below the pre-set threshold, a corresponding alarm is generated on the access device to warn the customer of the network quality.

The QoS alarm function is used to monitor three indexes, packet loss, loop delay, and jitter. The corresponding values can be set according to the actual network condition. During a call, the access device collects the data of packet loss, loop delay, and jitter, and then compares the data with the preset thresholds. When the data exceeds the thresholds, an alarm is generated. When the network indexes return to be lower than the preset thresholds, a recovery alarm is generated on the access device.

The QoS alarms can detect the network abnormalities in real time. When users complain, the QoS alarm can be referred to locate the fault (whether caused by the network or device).

1.17 Voice Reliability

This topic describes features related to voice reliability, including dual-homing networking, highly reliable transmission (SCTP), and voice QoS.

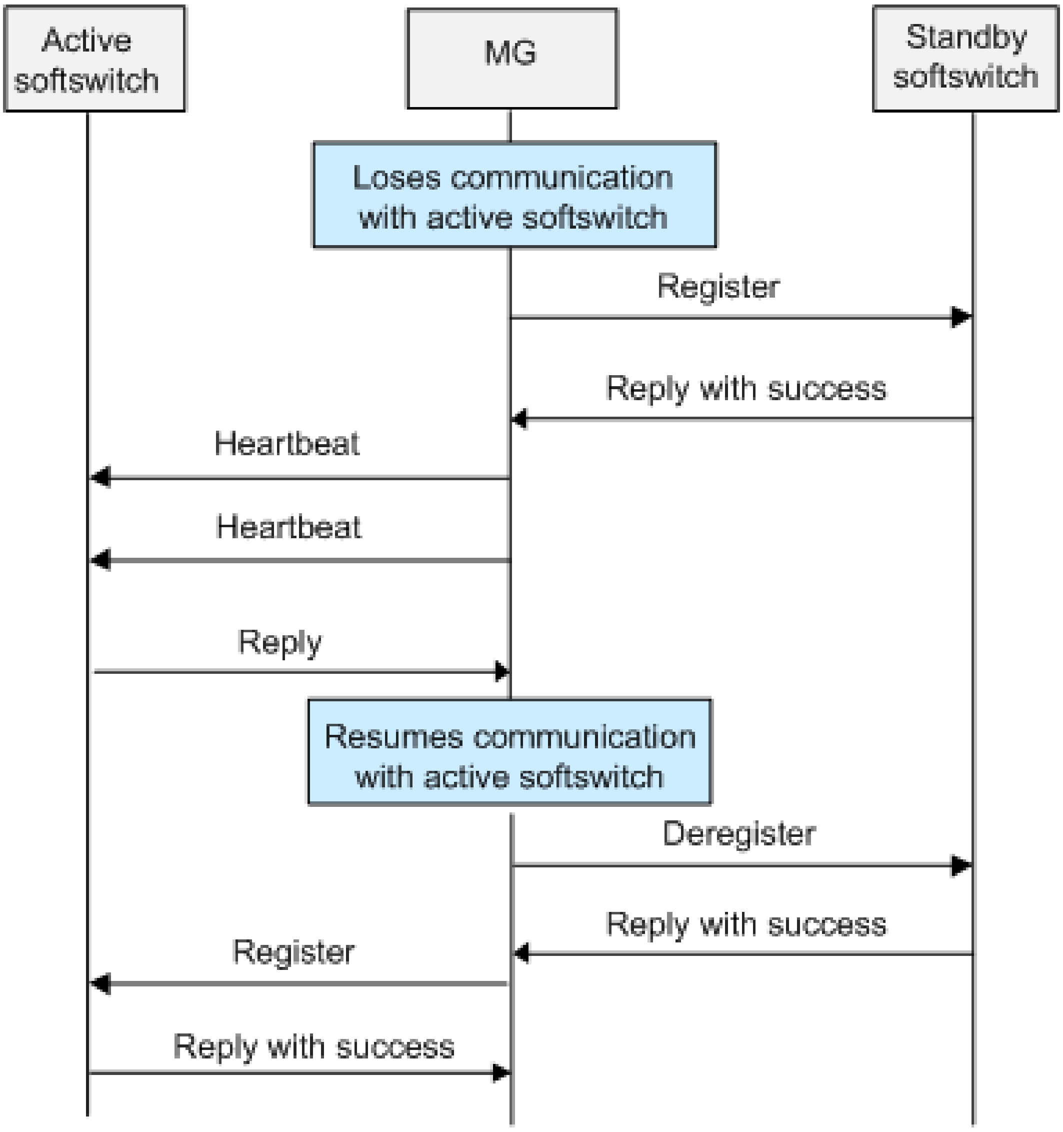
1.17.1 H.248/MGCP Dual Homing

Dual homing is an NGN (Next Generation Network) total solution. Based on this solution, when the active softswitch or the link from the MG to the active softswitch is faulty, the MG need be switched to the standby softswitch immediately to prevent call services of users connected to the softswitch and the MG from being affected.

Dual homing requires that one MG is configured with two softswitches, one active and one standby. The connection between the MG and the softswitch is detected through the heartbeat message.

Figure 1-158 illustrates the working principle of dual homing.

Figure 1-158 Working principle of dual homing



1. The MG detects the interrupted connection between the MG and the active softswitch through the heartbeat message.
2. The MG registers with the standby softswitch.
3. The MG sends the detection messages to the active softswitch at regular intervals (same as common heartbeat intervals), If the MG receives the response from the active softswitch, it indicates that the communication with the active softswitch is recovered. In this case, the MG takes the next action. If receiving no response from the softswitch, the MG keeps sending the detection messages.
4. The MG sends a message to the standby softswitch for service cancellation and waits for the response from the softswitch.
5. After receiving the response from the standby softswitch, the MG starts to register with the active softswitch. If three consecutive attempts of registration fail, the MG registers with the standby softswitch again following the same procedure.

Different carriers may choose the following different dual homing policies:

1. When the original active softswitch recovers, the MG automatically switches to the original active softswitch.
2. The MG does not support the auto-switching. Regardless of whether the MG registers with the active softswitch or the standby softswitch, if the softswitch with which the MG registers is normal, the MG works with this softswitch all along. The MA5600T/MA5603T/MA5608T can support the preceding two policies through related configuration. By default, the MA5600T/MA5603T/MA5608T supports the second policy.

1.17.2 H.248 Multi-homing

Overview

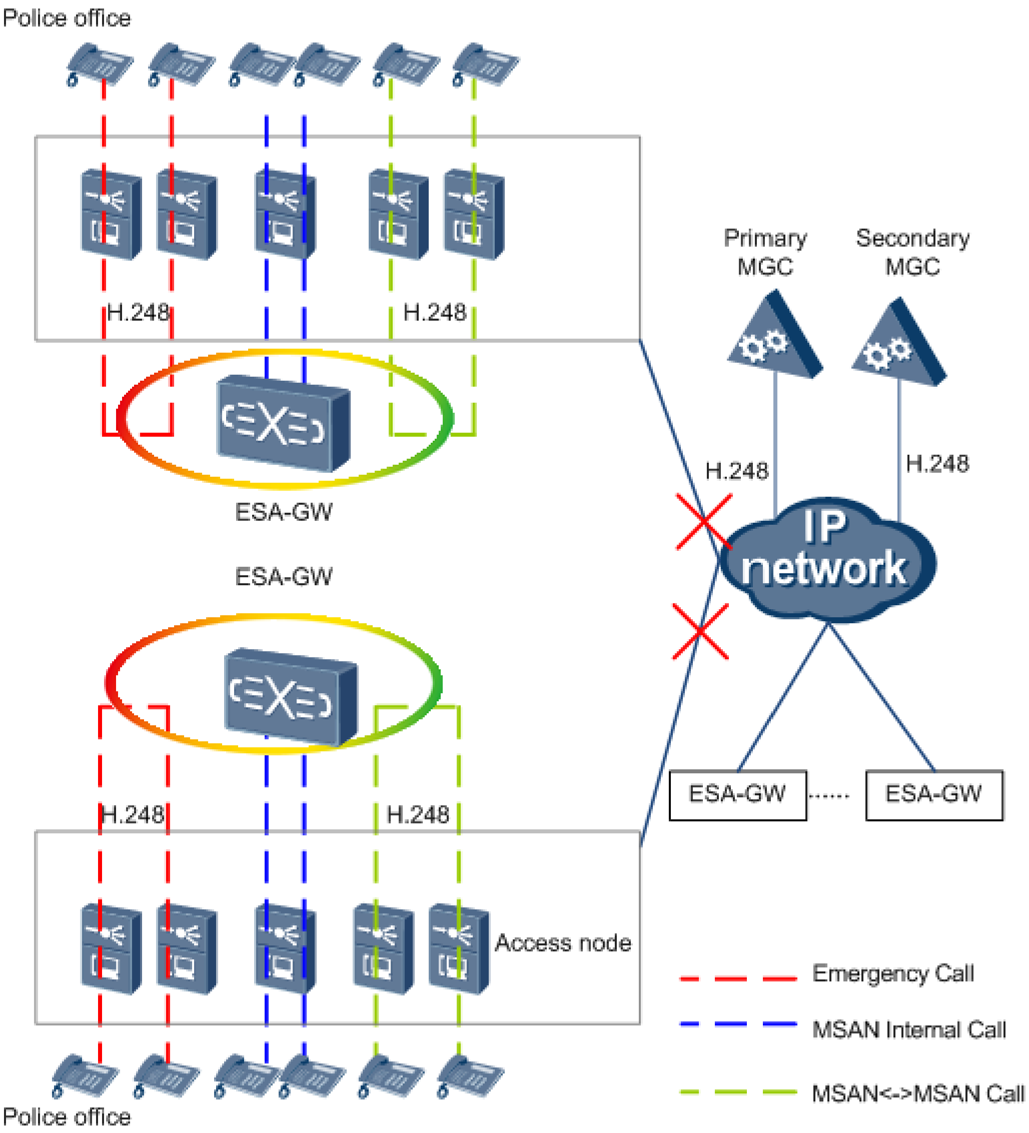
As an enhancement of dual-homing, multi-homing is a configuration in which a media gateway (MG) is homed to the primary media gateway controller (MGC), secondary MGC, and disaster-recovery MGC.

The system supports the following configurable switching policies for multi-homing:

1. Automatic switching back
   - An MG registering with the secondary MGC will automatically switch back to the primary MGC when the primary MGC recovers.
   - An MG registering with the disaster-recovery MGC will automatically switch back to the primary/secondary MGC when the primary/secondary recovers.
2. No automatic switching back
   - An MG registering with the secondary MGC will not automatically switch back to the primary MGC when the primary MGC recovers.
   - An MG registering with the disaster-recovery MGC will not automatically switch back to the primary/secondary MGC when the primary/secondary MGC recovers.

Network Application

Figure 1-159 H.248 multi-homing network



- ESA-GW: emergency standalone-gateway
- MSAN: multi-service access node

As shown in the preceding figure, an MSAN is an MA5600T/MA5603T/MA5608T and an ESA-GW can be a small-capacity softswitch in network deployment. Generally, ESA-GWs and MA5600T/MA5603T/MA5608Ts are deployed in the same telecommunications room. When disconnected from the primary and secondary MGCs (for example, due to a fiber cut), the MA5600T/MA5603T/MA5608T initiates registration to the ESA-GW. After the successful registration, all call services of the MA5600T/MA5603T/MA5608T are controlled by the ESA-GW. In this way, the following services are still available even if the MA5600T/MA5603T/MA5608T is disconnected from the primary and secondary MGCs:

1. Call services of users connected to the same MA5600T/MA5603T/MA5608T

2. Call services of users connected to different MA5600T/MA5603T/MA5608Ts (homed to the same ESA-GW)
3. Emergency call services

Pay attention to the following aspects when applying H.248 multi-homing:

1. The call service capabilities are restricted by the ESA-GW.
2. The callee of an emergency call (for example, police emergency call) must be connected to an MA5600T/MA5603T/MA5608T.

NOTE

This limitation is a supplement to the solution shown in Figure 1-159. Whether such an limitation takes effect depends on the actual core network topology.

3. Only the POTS users are supported (the ISDN users and other users are not supported).

Switching Process

Figure 1-160 Process of switching to the ESA-GW

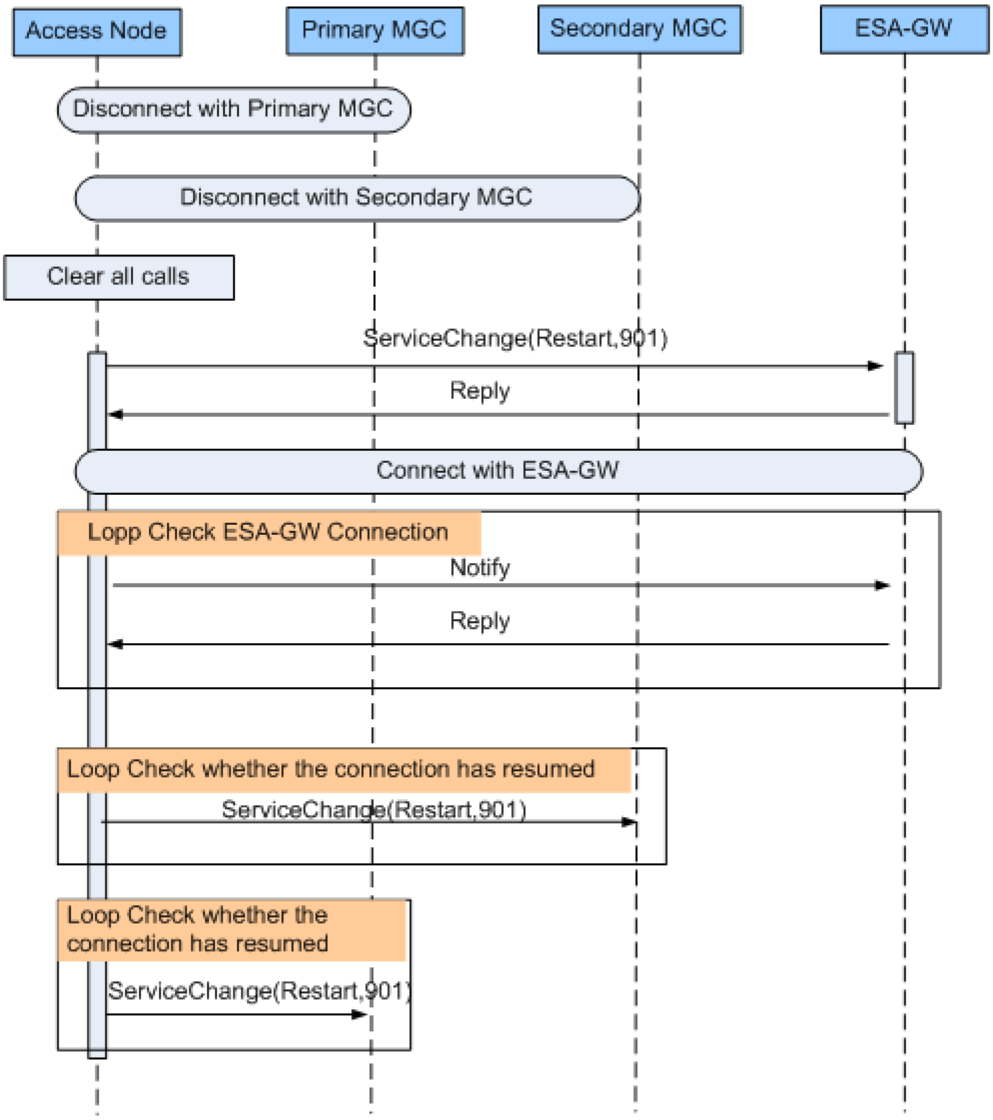
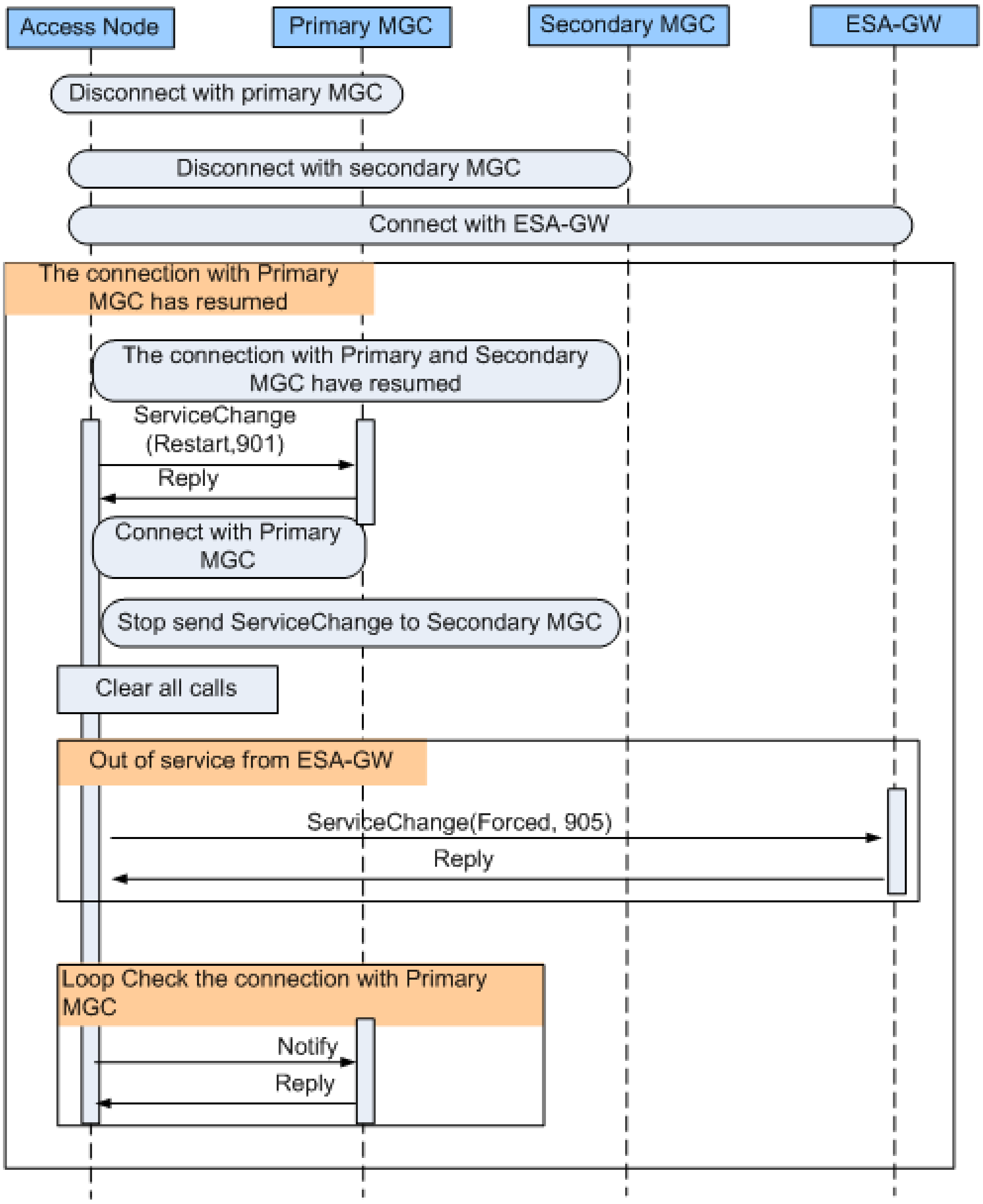

Access Node
Primary MGC
Secondary MGC
ESA-GW
Disconnect with Primary MGC
Disconnect with Secondary MGC
Clear all calls
ServiceChange(Restart,901)
Reply
Connect with ESA-GW
Lopp Check ESA-GW Connection
Notify
Reply
Loop Check whether the connection has resumed
ServiceChange(Restart,901)
Loop Check whether the connection has resumed
ServiceChange(Restart,901)

Figure 1-161 Process of switching back


Access Node
Primary MGC
Secondary MGC
ESA-GW
Disconnect with primary MGC
Disconnect with secondary MGC
Connect with ESA-GW
The connection with Primary MGC has resumed
The connection with Primary and Secondary MGC have resumed
ServiceChange (Restart,901)
Reply
Connect with Primary MGC
Stop send ServiceChange to Secondary MGC
Clear all calls
Out of service from ESA-GW
ServiceChange(Forced, 905)
Reply
Loop Check the connection with Primary MGC
Notify
Reply

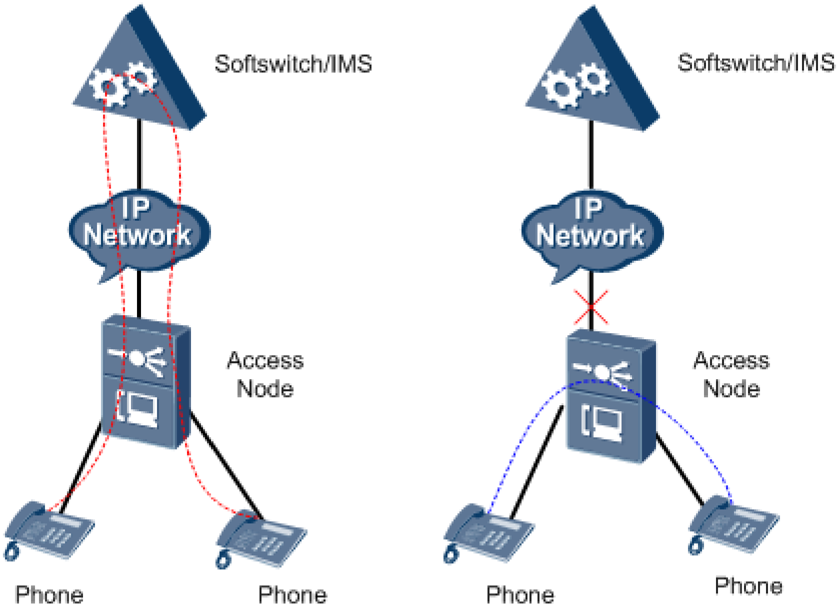
1.17.3 Emergency Standalone

Emergency standalone is a solution in which the users on the same MG can call each other even when the interface between the MG and the softswitch/IMS is interrupted. After a user picks up the telephone, the MG (namely, the MA5600T/MA5603T/MA5608T) checks whether the interface connected the softswitch/IMS is interrupted.

- If the interface is in the normal state, the normal softswitch/IMS process starts.
- Otherwise, the MG checks whether emergency standalone can be enabled.
  - If yes, the MA5600T/MA5603T/MA5608T controls the call process.
  - If no, the user listens to the busy sound (because the interface is faulty and emergency standalone is not allowed).

Figure 1-162 shows the operating principle of emergency standalone.

Figure 1-162 Operating principle of emergency standalone



The emergency standalone process is as follows:

- The service processing after the calling party picks up the telephone is as follows:
  a. The calling party picks up the telephone.
  b. The device automatically delivers the dial tone to the calling party.
  c. The calling party dials a phone number.
  d. The device analyzes the dialed number and finds out the called party on the same device. The phone number is configured for a user when the user is configured on the MG.
  e. The device delivers the ringing signal and calling party's phone number to the called party.
  f. The device delivers the ring-back tone to the calling party.
- The service processing after the called party picks up the telephone is as follows:
  a. The called party picks up the telephone.
  b. The device stops delivering the ring-back tone to the calling party.
  c. The calling and called parties start a conversation.
- The service processing after any party puts down the telephone is as follows:
  a. Any of the two parties puts down the telephone.
  b. The device delivers the busy tone to the other party.
  c. The other party puts down the telephone.

NOTE

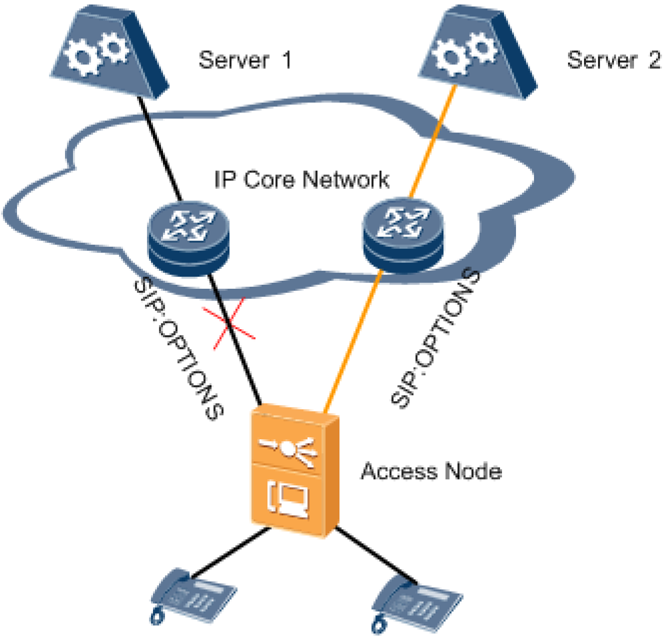
- Only the unabbreviated number is supported and the Centrex group, abbreviated number message, user outgoing/incoming call authority, and various new services are not supported.
- A user can call another user only on the same VAG.
- The feature applies only to the VoIP user.

- The dual-homing feature and the emergency standalone feature cannot be enabled at the same time.
- The ISDN services do not support the function of emergency standalone when it uses the H.248 protocol.

1.17.4 SIP Dual Homing

Figure 1-163 shows the networking of SIP dual homing.

Figure 1-163 Call releasing flow



The working flow of SIP dual homing is similar to the working flow of H.248/MGCP dual homing. The access device detects the proxy server in real time. When the primary proxy server is faulty, services can be switched to the secondary proxy server. Before the switching, the call can be released. After the switching, the call can be initiated.

1.17.5 H.248/SIP over SCTP

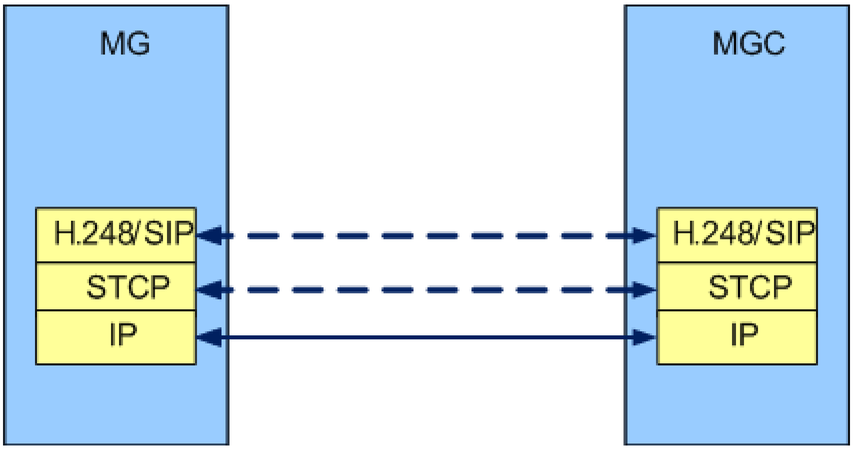
Currently, most devices adopt H.248/SIP over UDP. H.248.4 recommends H.248/SIP over SCTP, which implements the message retransmission at the application layer through SCTP.

Compared with the UDP protocol, the SCTP protocol has the following advantages:

1. Reliability: Messages can be transmitted fast and reliably through SCTP.
2. Multi-homing: With the multi-homing feature, multiple IP addresses are supported on an SCTP endpoint. That is, an SCTP endpoint can use multiple physical network ports to enhance the endpoint reliability.
3. Congestion control: The congestion control through SCTP is similar to the congestion control through TCP.
4. Heartbeat mechanism: SCTP provides the heartbeat mechanism at the network layer.
5. Security: Four-way handshake and cookie mechanisms effectively prevent the DoS attack.

As shown in Figure 1-164, the IP protocol is used at the network layer, SCTP the transport layer, and H.248/SIP the application layer.

Figure 1-164 Protocol architecture of H.248/SIP over STCP

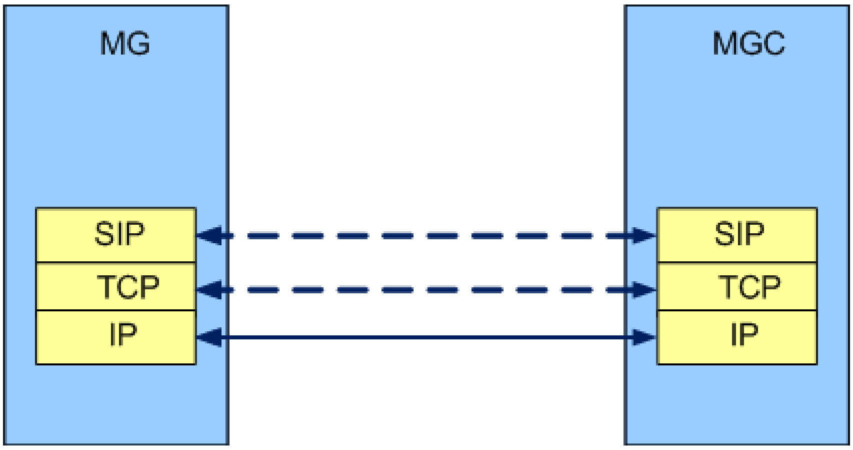


1.17.6 SIP over TCP

Some carriers require the TCP-based SIP signaling transmission, which implements the packetization of the SIP packet (the SIP packet is large in size) and enhances the transmission reliability through TCP.

As shown in Figure 1-165, the IP protocol is used at the network layer, TCP the transport layer, and SIP the application layer.

Figure 1-165 Protocol architecture of SIP over TCP



1.17.7 Voice QoS

The voice service requires high real-time performance, low delay, and fast call connection. Therefore, the voice packets should be forwarded with a high priority. The router, however, forwards the packets based on the VLAN priority (complying with 802.1p) and DSCP/ToS set in the packets.

802.1p Priority (Separately Set for Signaling and Media Streams)

Figure 1-166 802.1q frame format

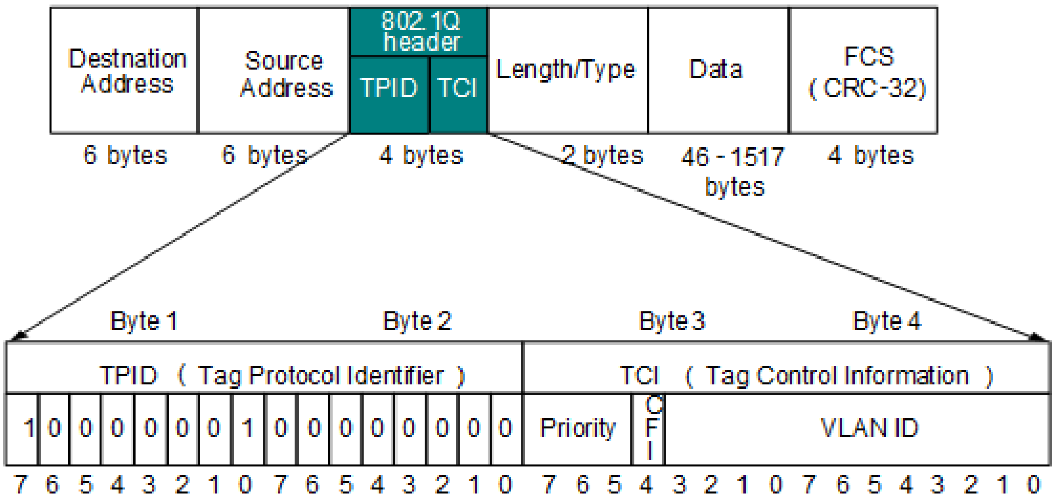


Figure 1-166 shows the Ethernet frame format defined in 802.1q. The four-byte 802.1q header contains the following contents:

- Tag protocol identifier (TPID): Two-byte tag protocol identifier, with the value of 8100.
- Tag control information (TCI): Two-byte tag control information. It is a new type of information defined by IEEE, indicating a text added with the 802.1q label. The TCI is divided into the following three fields:
  - VLAN ID: 12-bit, indicating the VLAN ID. Up to 4096 VLANs are supported. All the data packets transmitted from the host that supports 802.1q contain this field, indicating the VLAN to which the data packets belong.
  - Canonical format indicator (cfi): one-bit. It is used in the frame for data exchange between the Ethernet network of the bus type and the FDDI or token ring network.
  - Priority: three-bit, indicating the priority of the frame. Up to eight priorities are supported. It determines the data packet to be transmitted first in case of switch congestion.

The local media IP address and signaling IP address of the MA5600T/MA5603T/MA5608T can be configured in one VLAN or different VLANs according to the networking requirements. The 802.1p priorities (in the range of 0-7) can be set for the media IP address and signaling IP address respectively. By default, the priority for either the media IP address or the signaling IP address is 6.

DSCP/TOS

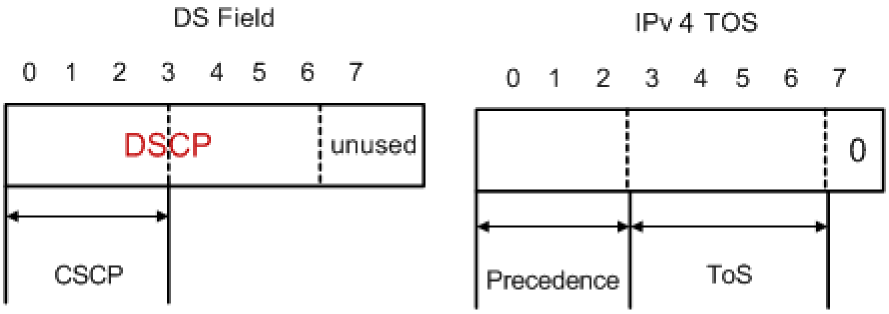
As defined in the IP protocol, the DSCP and ToS occupy the same field (one-byte) in the IP header. The device on the IP bearer network identifies whether DSCP or ToS is filled in the IP header, and schedules and forwards the packets with the DSCP/ToS field according to the settings to ensure the QoS for different services.

The type of service (ToS) field contains a three-bit precedence subfield (ignored currently), a four-bit ToS sub field, and one reserved bit (it must be set to 0). The four bits in the ToS sub field represent the minimum delay, maximum throughput, maximum reliability, and minimum cost respectively. Only one of the four bits can be set. If all four bits are set to 0, it indicates the common service.

The DSCP identification is based on the IPv4 ToS and the IPv6 traffic class.

As shown in Figure 1-167, the first six bits in the DS field (bits 0-5) are used to differentiate the DS codepoints (DSCPs) and the last two bits (bits 6 and 7) are reserved. The first three bits in the DS field (bits 0-2) are the class selector codepoint (CSCP), which indicates a class of DSCP.

Figure 1-167 DSCP identification format



DSCP is used to select the corresponding per-hop behavior (PHB) on all the nodes of the network. The PHB describes the external visible behaviors when the DS node functions on the data stream aggregation. Currently, IETF defines three types of PHB: expedited forwarding (EF), assured forwarding (AF), and best-effort. For example,

- BE: DSCP = 000000
- EF: DSCP = 101110
- The AF codepoints are as follows:

	Low Discard Priority, j = 1	Middle Discard Priority, j = 2	High Discard Priority, j = 3
AF (i = 4)	100010	100100	100110
AF (i = 3)	011010	011100	011110
AF (i = 2)	010010	010100	010110
AF (i = 1)	001010	001100	001110

The first three bits (bits 0-2) for one type of AFs are the same. To be specific, the first three bits of AF1 are 001, AF2 010, AF3 011, and AF4 100. Bits 3-4 represent the discard priority, namely, 01, 10, and 11. The larger the value, the higher the discard priority.

The DSCP/ToS value of local media IP packet and signaling IP packet can be configured on the MA5600T/MA5603T/MA5608T respectively. First the configuration policy (DSCP or ToS) is selected, and then the corresponding value is set. By default, DSCP is selected on the MA5600T/MA5603T/MA5608T, with the value of 56 (EF with the highest priority).

1.17.8 Emergency Call

The MA5600T/MA5603T/MA5608T identifies emergency calls according to emergency call digitmaps and ensures successful emergency calls by reserving digital signal processing (DSP) resources and implementing CPU overload control (OLC).

Definition

An emergency call is a kind of voice service initiated by a user using a terminal in case of emergencies. Generally, a fixed emergency call number is used within a country or region, for example, 911 in America and 110/119 in mainland China. When a user dials such a number, the system identifies the call as an emergency call, implements special processing on the call, and sends a request for routing the call to a specific emergency call center.

Device Implementation

The MA5600T/MA5603T/MA5608T identifies emergency calls according to emergency call digitmaps and ensures successful emergency calls by reserving DSP resources and implementing CPU OLC.

Digitmap

A digitmap is a set of digit collection descriptors. It is a dialing scheme resident in the MSAN and is used for detecting and reporting digit events received on a termination.

An emergency call digitmap is a dialing scheme for emergency calls. Using such digitmaps, emergency calls can be distinguished from common calls. The MSAN and core network devices, such as SoftX3000 or IMS, will then take protective measures for these emergency calls.

The MA5600T/MA5603T/MA5608T supports manual configuration of emergency call digitmaps, meeting requirements in different countries or regions.

NOTE

Digitmap configuration is complex because a digitmap is constituted by characters that have specific meanings and usage. Such information is defined in protocols and will not be detailed in this topic. It is recommended that you refer to the digitmap description in ITU-T H248.1 (applicable to the H.248 protocol) or ITU-T SIP.1 (applicable to the SIP protocol) before configuring a digitmap.

DSP Resource Reservation

DSP resources are the most important voice resources in VoIP and are used for converting TDM service flows into IP service flows. To ensure a high success rate of emergency calls, the system reserves some DSP resources (which are configurable) for callers and callees of emergency calls.

Figure 1-168 illustrates how the system reserves DSP resources for callers of emergency calls.

Figure 1-168 Reserving DSP resources for callers of emergency calls

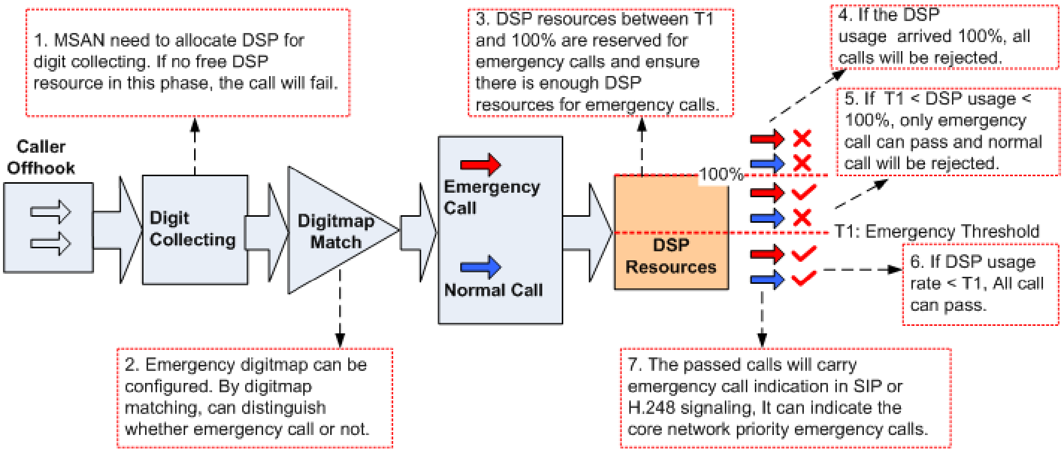
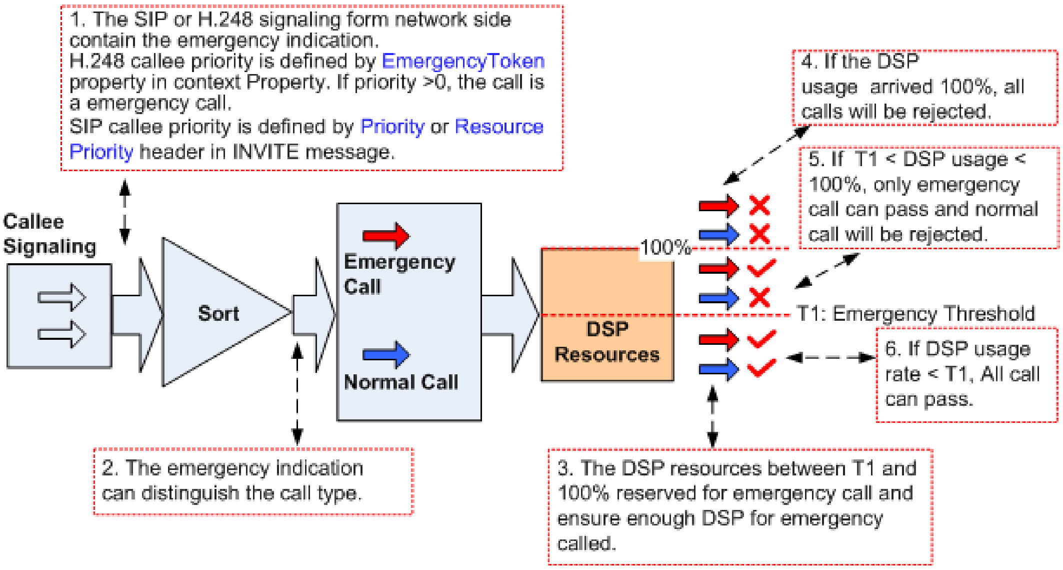

1. MSAN need to allocate DSP for digit collecting. If no free DSP resource in this phase, the call will fail.
3. DSP resources between T1 and 100% are reserved for emergency calls and ensure there is enough DSP resources for emergency calls.
4. If the DSP usage arrived 100%, all calls will be rejected.
5. If T1 < DSP usage < 100%, only emergency call can pass and normal call will be rejected.
Caller Offhook
Digit Collecting
Digitmap Match
Emergency Call
Normal Call
100%
DSP Resources
T1: Emergency Threshold
6. If DSP usage rate < T1, All call can pass.
2. Emergency digitmap can be configured. By digitmap matching, can distinguish whether emergency call or not.
7. The passed calls will carry emergency call indication in SIP or H.248 signaling, It can indicate the core network priority emergency calls.

Figure 1-169 illustrates how the system reserves DSP resources for callees of emergency calls.

Figure 1-169 Reserving DSP resources for callees of emergency calls


1. The SIP or H.248 signaling form network side contain the emergency indication.
H.248 callee priority is defined by EmergencyToken property in context Property. If priority >0, the call is a emergency call.
SIP callee priority is defined by Priority or Resource Priority header in INVITE message.
4. If the DSP usage arrived 100%, all calls will be rejected.
5. If T1 < DSP usage < 100%, only emergency call can pass and normal call will be rejected.
Callee Signaling
Sort
Emergency Call
Normal Call
100%
DSP Resources
T1: Emergency Threshold
6. If DSP usage rate < T1, All call can pass.
2. The emergency indication can distinguish the call type.
3. The DSP resources between T1 and 100% reserved for emergency call and ensure enough DSP for emergency called.

CPU OLC

CPU OLC prevents exhaustion of equipment CPU resources. It protects equipment from service interruption or NMS management failure that is triggered by CPU overload in case of heavy traffic. CPU OLC also ensures to a certain extent the quality of high priority services when the system is overloaded.

The system employs CPU OLC for callers and callees of emergency calls, ensuring a high success rate of emergency calls.

Figure 1-170 illustrates how the system implements CPU OLC for callers of emergency calls.

Figure 1-170 Implementing CPU OLC for callers of emergency calls

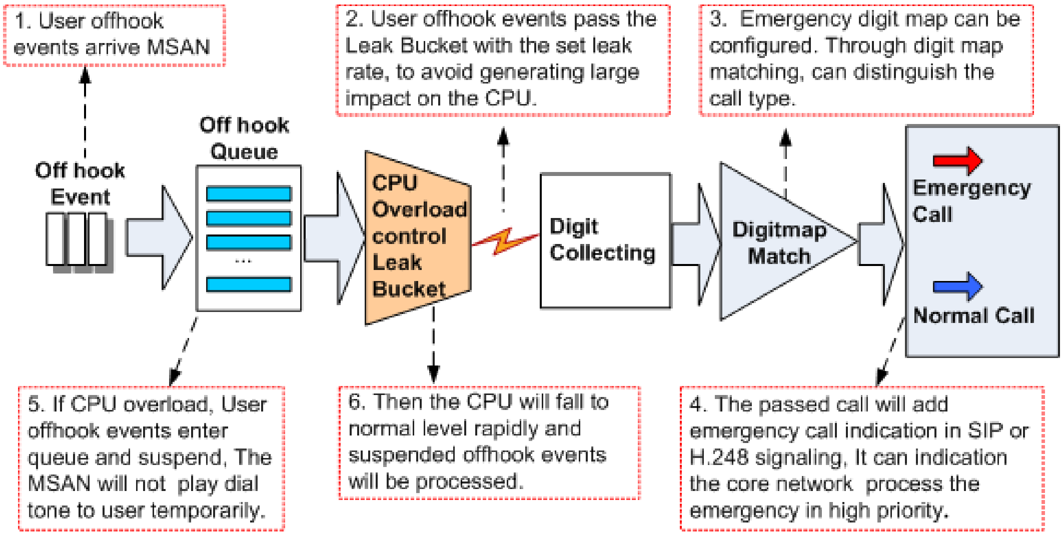
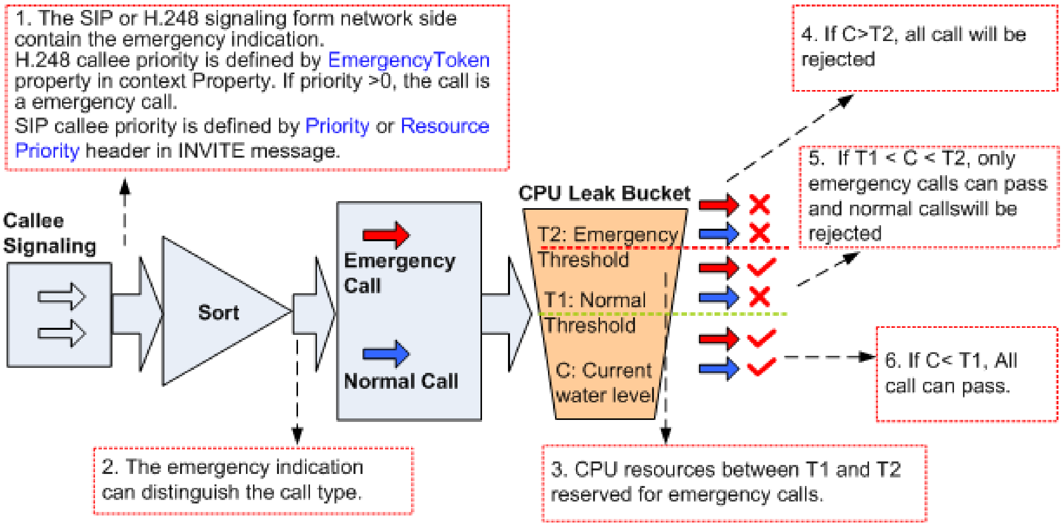

1. User offhook events arrive MSAN
2. User offhook events pass the Leak Bucket with the set leak rate, to avoid generating large impact on the CPU.
3. Emergency digit map can be configured. Through digit map matching, can distinguish the call type.
Off hook Queue
Off hook Event
...
CPU Overload control Leak Bucket
Digit Collecting
Digitmap Match
Emergency Call
Normal Call
5. If CPU overload, User offhook events enter queue and suspend, The MSAN will not play dial tone to user temporarily.
6. Then the CPU will fall to normal level rapidly and suspended offhook events will be processed.
4. The passed call will add emergency call indication in SIP or H.248 signaling, It can indication the core network process the emergency in high priority.

Figure 1-171 illustrates how the system implements CPU OLC for callees of emergency calls.

Figure 1-171 Implementing CPU OLC for callees of emergency calls


1. The SIP or H.248 signaling form network side contain the emergency indication.
H.248 callee priority is defined by EmergencyToken property in context Property. If priority >0, the call is a emergency call.
SIP callee priority is defined by Priority or Resource Priority header in INVITE message.
4. If C>T2, all call will be rejected
5. If T1 < C < T2, only emergency calls can pass and normal callswill be rejected
6. If C< T1, All call can pass.
Callee Signaling
Sort
Emergency Call
Normal Call
CPU Leak Bucket
T2: Emergency Threshold
T1: Normal Threshold
C: Current water level
✗
✗
✓
✗
✓
✓
2. The emergency indication can distinguish the call type.
3. CPU resources between T1 and T2 reserved for emergency calls.

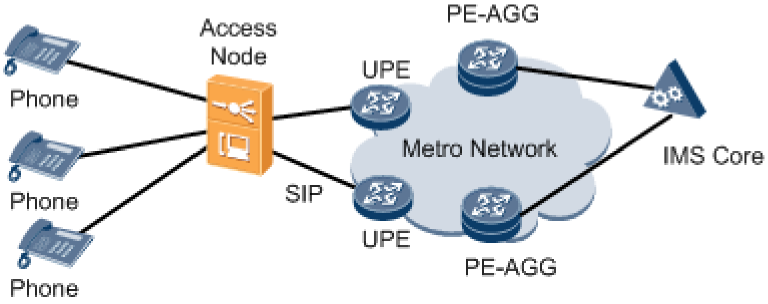
1.18 Configuring the VoIP PSTN Service (SIP-based)

The SIP-based VoIP technology makes the transport network evolve to the IP network without decreasing the voice quality, provides more value-added functions for users, and saves expense.

Application Context

As shown in Figure 1-172, the Access node functions as an SIP access gateway. In the downstream direction, it provides the access to PSTN users; in the upstream direction, it is connected to the IMS system, working with the IMS core to provide the VoIP service based on SIP.

Figure 1-172 Example network of the SIP voice service



Prerequisite

- The current system protocol is the SIP protocol. If the current system protocol is not the SIP protocol, change the current system protocol to the SIP protocol with reference to the 1.18.1.3 Adding an SIP Interface.
- According to the actual network, a route from the Access node to the IMS core network device must be configured to ensure that the Access node and the IMS core network device are reachable from each other.
- The voice daughter board on the control board works in the normal state.
- Electronic switch 1 must be in location-0 (indicating that the VoIP service is supported) If the SCUB control board is used. For details of the configuration method, see **electro-switch**.

Data preparation

Table 1-23 provides the data plan for configuring the VoIP service.

Table 1-23 Data plan for configuring the SIP-based VoIP service

Item			Remarks
SIP interface data (Must be the same as that on the IMS core network device.)	Parameters related to the media stream and the signaling flow	Media and signaling upstream VLAN	It is used as the upstream VLAN of the VoIP service to be configured. Note that the media stream and the signaling stream can use the same VLAN or different VLANs. The result is determined according to the negotiation with the upstream device.
		Signaling upstream port	Upstream port for configuring the SIP signaling.
		Media IP address and signaling IP address	These IP addresses must be selected from the media and signaling IP address pools. The media and signaling IP address pools consists of all the IP addresses of the Layer 3 interface of the media and signaling

Item			Remarks
			upstream VLAN.
		Default IP address of the MG	Next hop address from the Access node to the IMS core network device. **NOTICE** If the media IP address and the signaling IP address are different and the media and the signaling are transmitted upstream through different gateways, ensure that they correspond to correct gateways. Otherwise, normal calls may not be made.
	Parameters of the SIP interface **NOTE** Parameters listed here are mandatory, which means that the SIP interface fails to be enabled if these parameters are not configured.	SIP interface ID	It is the SIP interface ID used for the VoIP service to be configured, which determines the virtual access gateway (VAG) specified for the user.
		Signaling port ID of the SIP interface	The value range is 5000-5999. The protocol defines the port ID as 5060.
		IP address of the active IMS core network device to which the SIP interface belongs	When dual homing is not configured, parameters of only one IMS core network device are required. If dual homing is configured, the IP address and the port ID of the standby IMS core network device must be configured.
		Port ID of the active IMS core network device to which the SIP interface belongs	
		Transmission mode of the SIP interface	The transmission mode is selected according to the requirements on the IMS core network device. Generally, UDP is adopted.
		Home domain of the SIP interface	It corresponds to parameter **home-domain** in the MG interface attributes.
		Index of the profile used by the SIP interface	It corresponds to parameter **Profile-index** in the MG interface attributes.
		IP address obtaining mode of the proxy server	• In the IP mode, the IP address and the port ID of the active proxy server must be configured. • In the DNS-A or DNS-SRV mode, the domain of the active proxy server must be configured.
	Ringing mode of the SIP		According to the service requirements, the ringing mode of the SIP interface

Item			Remarks
	interface		is determined.
Voice user data (Must be the same as that on the IMS core network device.)	Slot of the voice service board		-
	User configuration data	Phone number	The phone number that the IMS core network device allocates to the user must be configured.
		User priority	According to the service requirements, user priority needs to be specified. The user priority includes the following: - cat1: government1 (category 1 government users) - cat2: government2 (category 2 government users) - cat3: common (common users)
		User type	According to the service requirements, user type needs to be specified. The user type includes the following: - DEL: direct exchange lines (default) - ECPBX: earth calling PBX - LCPBX: loop calling PBX - PayPhone: pay phone
	System parameters		The system parameters including the overseas version flag and message waiting indication (MWI) mode need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	Overseas parameters		The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	PSTN port attributes		If the PSTN port needs to support the polarity reversal accounting, the PSTN port needs to be configured to support the polarity reversal pulse. Other attributes do not need to be modified if there is no special requirement.
	Ringing current attributes		You can adjust the ringing volume by modifying the attributes of the ringing current. Generally, the parameters of the ringing current attributes do not

Item		Remarks
		need to be modified. You do not need to modify the parameters of the ringing current attributes according to the local standards only when the default ringing current attributes do not meet the local standards.

Procedure

1.18.1 Configuring an SIP Interface

As an interface used for the intercommunication between the MA5600T/MA5603T/MA5608T and the MGC, the interface is vital to the SIP-based VoIP service. Therefore, to implement the VoIP service, the SIP interface must be configured and must be in the normal state.

Procedure

1.18.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.18.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.18.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

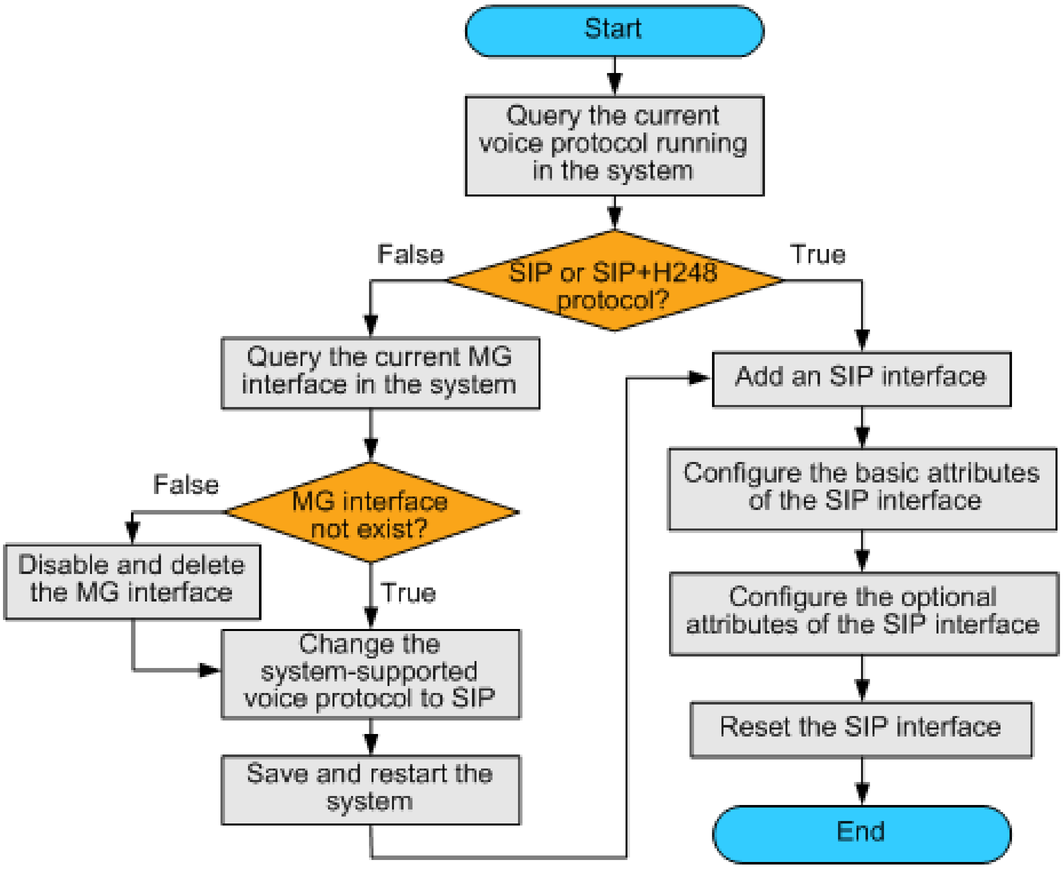
1.18.1.3 Adding an SIP Interface

The MA5600T/MA5603T/MA5608T exchanges the signaling and protocol packets with the information management system (IMS) through an SIP interface. To ensure uninterrupted signaling and protocol packet exchange, ensure that the status of the SIP interface is in a normal state after you add it.

Context

- One MA5600T/MA5603T/MA5608T supports up to eight SIP interfaces. Each SIP interface can be configured with the interface attributes separately.
- The SIP attributes configured for an SIP interface take effect on this interface only.
- The attributes of an SIP interface, including the signaling IP address, media IP address, transport-layer protocol, port ID of the transport-layer protocol, IP address (or domain name) of the proxy server, port ID of the proxy server, home domain name, profile index, are mandatory. In addition, the attributes of an SIP interface must be consistent with those configured on the IMS side so that the status of the SIP interface is normal.

Configuration Flowchart


Start
Query the current voice protocol running in the system
SIP or SIP+H248 protocol?
False
True
Query the current MG interface in the system
MG interface not exist?
False
True
Disable and delete the MG interface
Change the system-supported voice protocol to SIP
Save and restart the system
Add an SIP interface
Configure the basic attributes of the SIP interface
Configure the optional attributes of the SIP interface
Reset the SIP interface
End

Procedure

Step 1 Query the current voice protocol running in the system.

Run the **display protocol support** command to query the voice protocol that is currently supported by the system.

- If the system voice protocol is the SIP protocol, go to Step 6.
- If the system voice protocol is not the SIP protocol, go to Step 2.

Step 2 Query the current MG interface in the system.

Run the **display if-mgcp all** or **display if-h248 all** command to query whether an MGCP interface or an H.248 interface currently exists in the system.

- If there is no such an MG interface, go to Step 4.
- If there is such an MG interface, go to Step 3.

Step 3 Disable and delete the MG interface.

1. Delete the user data of this MG interface, and then run the **shutdown(mgcp)** or **shutdown(h248)** command to disable the MG interface according to the protocol type of the interface.

NOTICE

This operation interrupts all the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation. Before performing this operation, you must check whether the operation is allowed.

2. Run the **undo interface mgcp** or **undo interface h248** command to delete the MG interface.

Step 4 Change the system-supported voice protocol to SIP.

Run the **protocol support** command to change the system-supported voice protocol to SIP.

Step 5 Save the configuration data and restart the system.

Save the configuration data by running the **save** command. Then run the **reboot** command to restart the system to make the new configuration data take effect.

Step 6 Run the **interface sip** command to add an SIP interface.

Step 7 Run the **if-sip attribute basic** command to configure the basic attributes of the SIP interface.

NOTE

- Between **proxy IP** and **proxy domain**, which are basic attributes, at least one attribute must be configured. If both attributes are configured, the system determines which attribute is used according to the configured address mode of the proxy server.
- The profile index must be configured.

Step 8 Run the **if-sip attribute optional** command to configure the optional attributes of the SIP interface.

The optional attributes include some new service types supported by the SIP interface, such as the conference call, message waiting indicator, UA-profile subscription, and REG-state subscription, and also include the SIP proxy configuration mode. The optional attributes require the IMS-side support, and must be consistent with those on the IMA-side.

After the configuration is completed, run the **display if-sip attribute config** command to query the attributes of the SIP interface.

Step 9 Reset the SIP interface.

Run the **reset** command to reset the SIP interface for the new configuration data to take effect. Otherwise, the configuration data does not take effect but is only stored in the database.

After the SIP interface is reset successfully, run the **display if-sip attribute running** command to query the running status of the SIP interface. If the active (or standby) proxy is **up**, the SIP interface is normal.

----End

Example

Assume that: the media IP address is 10.10.10.13, signaling IP address is 10.10.10.13, transmission protocol is UDP, port ID is 5000, active proxy server IP address 1 is 10.10.10.14, port ID of the active proxy server is 5060, active proxy server domain name is proxy.domain, standby proxy server IP address 1 is 10.10.10.15, port ID of the standby proxy server is 5060, home domain name is sip.huawei.com, and profile index is 1. To configure the attributes of SIP interface 0, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#if-sip attribute basic media-ip 10.10.10.13 signal-ip 10
.10.10.13 signal-port 5000 transfer udp primary-proxy-ip1 10.10.10.14 primary-pr
oxy-port 5060 primary-proxy-domain proxy.domain secondary-proxy-ip1 10.10.10.15
secondary-proxy-port 5060 home-domain sip.huawei.com sipprofile-index 1
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#
  Resetting SIP interface 0 succeeded
huawei(config-if-sip-0)#display if-sip attribute running
  ------------------------------------------------------------
  ...//The rest information in response to this command is omitted.
  Primary Proxy State               up   //Indicates that the SIP interface is in the
normal state.
  Secondary Proxy State              down
  ...
  ------------------------------------------------------------
```

1.18.1.4 (Optional) Configuring the Software Parameters of the SIP Interface

The software parameters of a SIP interface mainly define certain common service attributes of the SIP interface. After the software parameter configuration, the parameters take effect immediately and are valid only to the SIP interface. Skip this topic if the system default configuration meets user requirements.

Prerequisites

The MG interface has been configured. For details about how to configure the MG interface, see 1.18.1 Configuring an SIP Interface.

Context

The details about the software parameters that can be configured of a SIP interface that supports SIP, see section **Usage Guidelines** in **mg-software parameter**. The other parameters are reserved in the system.

Table 1-24 lists parameters that are usually configured to a non-default value. The other parameters are not required.

Table 1-24 Software parameters usually configured of a SIP interface

Parameter	Description	Default Setting
2	Indicates whether the standalone mode is supported.	Numeral type. Range: 0-1. • 0: indicates that the standalone function is not supported. • 1: indicates that the standalone function is supported. Default: 0 This parameter is usually set to **1**.
8	Indicates whether the heartbeat message of the MG is disabled.	Numeral type. Range: 0-1. • 0: the heartbeat message of the MG is disabled • 1: the heartbeat message of the MG is enabled Default value: 0.

Procedure

Step 1 Enter the SIP interface mode.

In global config mode, run the **interface sip** command to enter the SIP interface mode.

Step 2 Configure software parameters.

Run the **mg-software parameter** command the software parameters required in the data plan.

Step 3 Check whether the software parameters are the same as those in the data plan.

Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 2 of the SIP interface 0 to 1 so that the SIP interface supports standalone, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#mg-software parameter 2 1
huawei(config-if-sip-0)#display mg-software parameter 2
  ------------------------------------------------
  MGID:0         para index:2  value:1
  ------------------------------------------------
```

```
APPENDIX:
 -------------------------------------------------
Parameter Index:  Interface software parameter name:
   2 : SAL Support
       0: No
       1: Yes
```

1.18.1.5 (Optional) Configuring the Ringing Mode of the SIP Interface

This topic describes how to configure the ringing mode of the SIP interface to support the break-make ratios of various new ringing modes and make the ringing mode meet the local standards.

Prerequisites

The SIP interface must be added successfully.

Context

- If the preset ringing modes of the system can meet the user requirements, you can select the required ringing mode and configure the corresponding ringing mapping.
- If the system-defined ringing modes cannot meet the user requirements, you can use the user-defined ringing mode and configure the corresponding ringing mapping.
- The user can configure the cadence duration for the user-defined ringing to form different ringing modes.
- The user-defined ringing modes are 0-15, which correspond to the cadence ringing modes 128-143 and initial ringing modes 144-159 defined by the user. For example, if the user-defined cadence ringing mode is 128, user-defined ringing mode 0 is selected. If the user-defined initial ringing mode is 144, user-defined ringing mode 0 is selected.
- The system supports up to 16 records of the ringing mode mapping.

Precaution

- The ringing mapping name must be unique on the same SIP interface.
- An index can be used for adding only one ringing mode on the same SIP interface.
- The 16 user-defined ringing modes can be modified but cannot be added.

Procedure

Step 1 According to the Usage Guidelines of the **ringmode add** command, check whether the preset ringing mode in the system meets the requirement.

- If the requirement is met, proceed to **step 4**.
- If the requirement is not met, go to **step 2**.

Step 2 In the global config mode, run the **user defined-ring modify** command to configure the user-defined ringing mode.

NOTE

- To use the user-defined ringing mode, perform this step and you can define the ringing types numbered 0-15.

- After configuring the user-defined ringing mode, you need to reset the corresponding service board to make the configuration data take effect, so that the user of the service board can use the new user-defined ringing mode. If the service board has been in service for a period, evaluate the impact on the online users before resetting the service board, and then determine whether to perform this operation.

Step 3 Run the **display user defined-ring** command to query the user-defined ringing.

Step 4 Run the **interface sip** command to enter the SIP mode.

Step 5 Run the **ringmode add** command to add a ringing mapping.

Run this command to configure the ringing mode for the users of the same SIP interface. The key parameters are described as follows:

- cadencering: Indicates the cadence ringing mode. The range 128-143 of this parameter corresponds to user-defined ringing modes 0-15.
- initialring: Indicates the initial ringing mode. The range 144-159 of this parameter corresponds to user-defined ringing modes 0-15.

Step 6 Run the **display user defined-ring** command to query ringing mapping records.

----End

Example

To add such a ringing mode mapping record on SIP interface 0, assume that:

- Index of the ringing mode mapping record: 1
- Name of the ringing mode mapping record: alert-group
- Cadence ringing mode: 1
- Initial ringing mode: 4

do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#ringmode add 1 alter-group cadencering 1 initialring 4
huawei(config-if-sip-0)#display ringmode 1
  ------------------------------------------------------------------
  MGID: 0
  Index: 1
  Ringmode-name: alter-group
  CadenceRing: Special Ring 1:2
  InitialRing: Normal Ring (FSK) 1:4
  ------------------------------------------------------------------
```

1.18.1.6 (Optional) Configuring User-Defined Signals

This topic describes how to configure user-defined signals on an SIP interface when the system default signals cannot meet customer requirements.

Prerequisites

The SIP interface is added successfully.

Context

Signals indicate the combination of different physical signals that are generated when the system needs to inform users of the call progress and call information. Physical signals include 3 types: media signals, line signals, and data signals.

- Media signals, including various signal tones (such as dial tone and busy tone), are generated by DSP chips.
- Line signals (such as ringing and polarity reversal signals) are generated by POTS boards.
- Data signals include calling line identification presentation (CLIP) signals, message waiting indicator (MWI) signals, and call cost signals. DSP chips process received pulse signals and then generate data signals.

When a signal is mapped to a specific scenario, the system generates this signal to inform users once this scenario occurs.

A signal may consist of a single signal unit. Figure 1-173 shows the signal configured for the scenario in which a user picks up the phone but does not dial a number for a long time. This signal consists of 2 signal units busy tone and howler tone. Table 1-25 lists signal attributes. For details, see"Parameter" of the **signal-unit set** command.

Figure 1-173 Signal configured for the scenario in which a user picks up the phone but does not dial a number for a long time

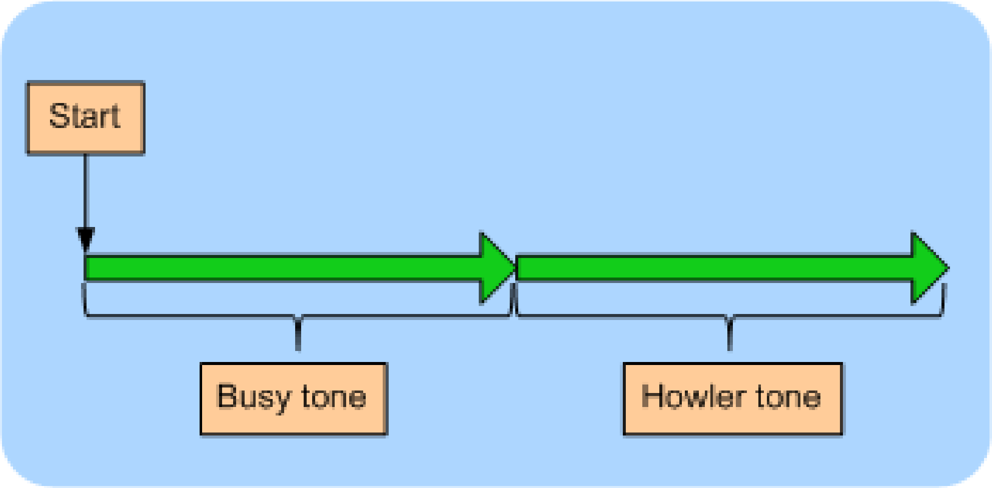


Table 1-25 Signal attributes

Attribute	Description
type	Indicates the type of the signal unit.
repeat	Indicates how many times this signal unit will be played.
duration	Indicates the duration of the signal unit. If it is set to 4294967295 (0xffffffff), the termination of signal unit playing depends on the termination condition.

Attribute	Description
start-condition	Indicates the condition for starting a signal unit.
end-condition	Indicates the condition for terminating a signal unit. If it is set to "-", this signal unit is continuously played and does not end.

Procedure

Step 1 Run the **interface sip** command to enter the SIP mode.

Step 2 Run the **display signal-scene** command to query whether the system default signals meet customer requirements. If the system default signals do not meet customer requirements, configure user-defined signal by referring to **Step 3**.

Step 3 In the SIP mode, run the **signal add** command to add user-defined signals.

NOTE

The name of a user-defined signal must be unique in a system. You can run the **display signal** command to query the signals existing in the system.

Step 4 Run the **signal-unit set** command to configure signal attributes.

Step 5 Run the **signal-mapping add** command to configure the mapping between the signal and scenario.

Step 6 Run the **reset** command to reset a SIP interface for the new configuration data to take effect.

NOTE

After the SIP interface is reset successfully, you can run the **display signal-scene** command to query the new configuration data.

----End

Example

Example 1: On SIP interface 0, add a new signal named signal1 for the scenario in which a user picks up the phone but does not dial a number for a long time. This signal consists of signal unit 0 and signal unit 1.

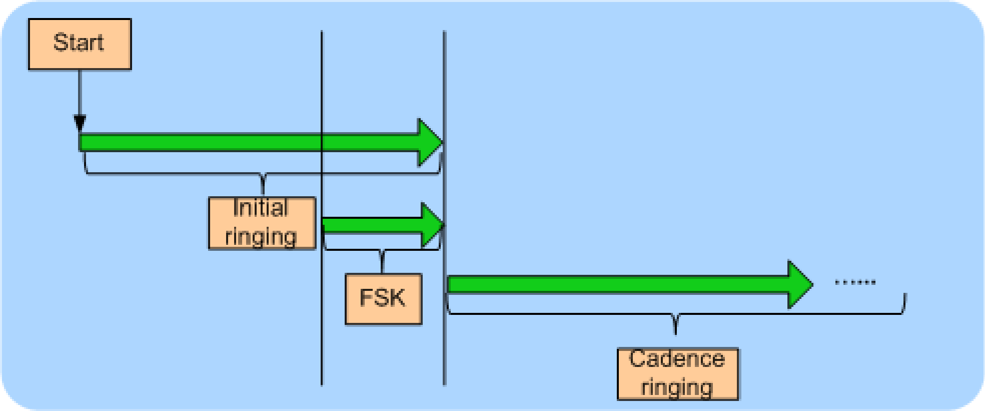
- Type of signal unit 0: tone 1 (busy tone). Signal unit 1 will be continuously played for 40s.
- Type of signal unit 1: tone 2 (howler tone). Signal unit 2 will be continuously played for 60s.

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#signal add signal1
huawei(config-if-sip-0)#signal-unit set signal1 0 type tone:1 duration 40000
huawei(config-if-sip-0)#signal-unit set signal1 1 type tone:2 duration 60000
huawei(config-if-sip-0)#signal-mapping add local:no_dial signal1
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#
  Resetting SIP interface 0 succeeded
```

Example 2: On SIP interface 0, add a new signal named signal2 for the scenario in which a user receives a normal incoming call. Figure 1-174 shows the signal configured for this scenario. This signal consists of signal unit 0, signal unit 1, and signal unit 2.

- Type of signal unit 0: init 0 (initial ringing). The initial ringing ends when signal unit 1 ends.
- Type of signal unit 1: data 0 (FSK CLIP).
  - Condition for starting signal unit 1: 650 ms elapse after the fsk_start signal is received or signal unit 0 is played for 4000 ms.
  - Condition for terminating signal unit 1: 350 ms elapse after the fsk_end signal is received. If the fsk_end signal is not received in a period of 3500 ms, signal unit 1 is automatically terminated.
- Type of signal unit 2: cade 0 (cadence ringing). When signal unit 0 ends, the system starts to play signal unit 2.

Figure 1-174 Signal2 configured for the scenario in which a user receives a normal incoming call



```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#signal add signal2
huawei(config-if-sip-0)#signal-unit set signal2 0 type init:0 end-condition 1e
huawei(config-if-sip-0)#signal-unit set signal2 1 start-condition fs:650|0s:4000
duration 3500
end-condition fe:350
huawei(config-if-sip-0)#signal-unit set signal2 2 type cade:0 start-condition 0e
huawei(config-if-sip-0)#signal-mapping add local:call_incoming signal2
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#
  Resetting SIP interface 0 succeeded
```

1.18.2 Configuring the VoIP PSTN User

After an SIP interface is configured, you can add plain old telephone service (POTS) users on the SIP interface to implement the VoIP PSTN service.

Procedure

1.18.2.1 Configuring the PSTN User Data

This topic describes how to configure the PSTN user data (the same as the corresponding data on the IMS) on the SIP interface to provide the POTS terminal with the access to the network for VoIP service.

Prerequisites

The POTS service board must be inserted into the planned slot, and the RUN ALM LED of the board must be green and must be on for 1s and off for 1s repeatedly.

NOTE

You can add a service board in two ways (see the Usage Guideline of the **board add** command). It is recommended that you insert the service board into the planned slot and then confirm the board.

Procedure

Step 1 In the global config mode, run the **board confirm** command to confirm the service board.

Step 2 Add a PSTN user.

1. In the global config mode, run the **esl user** command to enter the ESL user mode.
2. Run the **sippstnuser add** or **sippstnuser batadd** command to add the PSTN user.

NOTICE

When adding a user, you can configure the phone number (parameter **telno**). When the public ID is generated by the phone number, you must enter the phone number. It is recommended that you configure this phone number the same as the phone number configured on the IMS. In addition, ensure that the phone number is unique inside the AG.

3. Run the **sippstnuser auth set** command to configure the authentication data of the PSTN user.

NOTE

Considering users safety, the IMS may require user authentication. You can run the **sippstnuser auth set** command to configure the user authentication data, including user name, password mode and password. The authentication data should be consistent with that of IMS side.

4. Run the **display sippstnuser** command to check whether the PSTN user data is the same as that in the data plan.

Step 3 (Optional) Configure the attributes of the PSTN user.

The attributes of a PSTN user need to be configured when the default configuration is not consistent with the actual application.

1. Run the **sippstnuser attribute set** or **sippstnuser attribute batset** command to configure the attributes of the PSTN user.
2. Run the **display sippstnuser attribute** command to check whether the attributes of the PSTN user are the same as those in the data plan.

----End

Example

Assume that the ASPB service board is located in slot 0/2. To configure the attributes of the 64 SIP PSTN users (phone numbers are from 83110000 to 83110063) connected to SIP interface 0, set the PSTN user type of ports from 0/2/0 to 0/2/31 to payphone, the call priorities of PSTN users from ports 0/2/32 to 0/2/63 to Cat2, and use default values for other parameters, do as follows:

```
huawei(config-esl-user)#sippstnuser batadd 0/2/0 0/2/63 0 telno 83110000 step 1
huawei(config-esl-user)#sippstnuser attribute batset 0/2/0 0/2/31 potslinetype p
ayphone
huawei(config-esl-user)#sippstnuser attribute batset 0/2/32 0/2/63 priority cat2
```

1.18.2.2 (Optional) Configuring Digitmap for SIP Interfaces

The digitmap, also called number list, refers to the dialing plan on the access gateway (AG), which is used to detect and report dialing events received at the termination point. The digitmap defines number collection rules. It allows dialing events to be reported by groups, which reduces signaling exchanges between the AG and IMS.

Prerequisites

NOTICE

The digitmap configuration is relatively complicated. The information such as the meanings and usage of the characters in a digitmap is defined in the protocol, and is not described here. This topic provides only some basic information. You are advised to refer to digitmap description in SIP standard before configuring a digitmap.

Context

- Different digitmaps are required for different services. A digitmap group includes different digitmaps, providing customized digitmaps to accommodate to users' requirements. In this way, signaling exchanges are reduced between the AG and IMS.
- A digitmap consists of digit and character strings. When the received dialing sequence matches one of the character strings, you can infer that all numbers are received.
- The priority sequence of the digitmap is: user digitmap group > interface digitmap group > global local digitmaps. If a digitmap group used by a user does not have corresponding digitmaps, this user does not have the corresponding digitmaps. For example, digitmap group A is configured in user attributes, and digitmap group B is configured in the interface of the user. Besides, two-stage out-group digitmaps are not specified in digitmap group A, but two-stage out-group digitmaps are specified in digitmap group B. When digitmaps are used, the user does not load any two-stage out-group digitmaps because digitmap group A with a highest priority does not have two-stage out-group digitmaps (although two-stage out-group digitmaps are specified in digitmap group B and local digitmaps have two-stage out-group digitmaps). If the user cannot find any user-level or interface-level digitmap groups, the user uses global local digitmaps.
- If digitmaps are not configured, the system provides a default digitmap for the user, in which all telephone numbers can be matched.

Table 1-26 provides the characters defined in the SIP protocol for digitmaps. For details, refer to the SIP standard, which provides a better guide to the digitmap configuration.

Table 1-26 SIP digitmap format

Digit or Character	Description
0-9	Indicates dialed digits 0-9.
A-D	-
E	Indicates the asterisk (*) in dual tone multiple frequency (DTMF) mode.
F	Indicates the pound key (#) in DTMF mode.
X	Indicates a wildcard, which is a digit ranging from 0 to 9.
S	Indicates the short timer. After the timer times out; that is, the dialing plan matching is complete, the system reports numbers one by one if numbers remain.
L	Indicates the long timer. After the timer times out; that is, the dialing plan matching is complete, the system reports numbers one by one if numbers remain.
Z	Indicates duration modifier, which is a dialing event with a long duration. The dialing event is located in front of the event symbol with a specified position. When the duration of the dialing event exceeds the threshold, the dialing event satisfies this position.
.	Indicates that 0 or multiple digits or characters can exist before this character.
|	Is used to isolate character strings. Each character string is a selectable dialing plan.
[]	Indicates that one of the digits or characters in the square bracket is selected.

Procedure

- Configure a digitmap.
  a. In global configuration mode, run the **ocal-digitmap add** command to add a local preset digitmap.
  b. (Optional) In SIP mode, run the **digitmap-timer(sip)** command to configure a digitmap timer.
- Configure a digitmap group.
  a. In global configuration mode, run the **local-digitmap add** command to add a local preset digitmap.
  b. (Optional) In SIP mode, run the **digitmap-timer(sip)** command to configure a digitmap timer.
  c. Run the **local-digitmap-group add** command to add a digitmap group.

d. Run the **local-digitmap-group include** command to add local digitmap members to the digitmap group.

   The new digitmap group takes effect only when the user uses it in the next call.

e. Run the **mg-digitmap-group** command to configure the digitmap group used by the interface. The new digitmap group takes effect only when the user uses it in the next call.

f. Run the **sippstnuser attribute set** command to configure the digitmap group used by the user. The new digitmap group takes effect only when the user uses it in the next call.

----End

Example

For example, according to the data plan, digitmap group 1 is applied to users connected to the 0/6/0 port in the SIP interface. The digitmap group includes normal digitmaps and emergency digitmaps, whose formats are 8882xxxx and 8000xxxx respectively.

```
huawei(config)#local-digitmap add huawei normal 8882xxxx sip
huawei(config)#local-digitmap add huawei1 emergency 8000xxxx sip
huawei(config)#local-digitmap-group add DigitmapGroup1
huawei(config)#local-digitmap-group include DigitmapGroup1 huawei
huawei(config)#local-digitmap-group include DigitmapGroup1 huawei1
huawei(config)#interface sip 1
huawei(config-if-sip-1)#mg-digitmap-group DigitmapGroup1
huawei(config-if-sip-1)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser attribute set 0/6/0 cliptransseq digitmap-group
DigitmapGroup1
```

1.18.2.3 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  -------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
 ------------------------------------------------------------------------------
```

1.18.2.4 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1

  Command:
          display oversea parameters 1
    ------------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 100
  Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
  ------------------------------------------------------------------------------
```

1.18.2.5 (Optional) Configuring the Attributes of a PSTN Port

This topic describes how to configure the attributes of a PSTN port to ensure that the PSTN port can meet the actual application requirements.

Context

The MA5600T/MA5603T/MA5608T supports the following attributes of a PSTN port:

- Physical attributes (including whether to support the polarity reversal pulse, whether to support the port locking, and dialing mode). For details about how to configure the physical attributes of a PSTN port, see **pstnport attribute set**.
- Electrical attributes (including the impedance and the current). For details about how to configure the electrical attributes of a PSTN port, see **pstnport electric set**.
- KC attributes (including the KC accounting mode and the valid voltage). For details about how to configure the KC attributes of a PSTN port, see **pstnport kc set**.

Procedure

Step 1 In the global config mode, run the **pstnport** command to enter the PSTN port mode.

Step 2 Run the **pstnport attribute batset** or **pstnport attribute set** command to configure the physical attributes of the PSTN port.

Step 3 Run the **pstnport electric batset** or **pstnport electric set** command to configure the electrical attributes of the PSTN port.

Step 4 Run the **pstnport kc batset** or **pstnport kc set** command to configure the KC attributes of the PSTN port.

Step 5 Check whether the attribute configuration of the PSTN port is the same as that in the data plan.

- Run the **display pstnport attribute** command to query the physical attributes of the PSTN port.
- Run the **display pstnport electric** command to query the electrical attributes of the PSTN port.
- Run the **display pstnport kc** command to query the KC attributes of the PSTN port.

----End

Example

To configure the 32 PSTN ports of the board in slot 0/3 to support the polarity reversal accounting, do as follows:

```
huawei(config)#pstnport
huawei(config-pstnport)#pstnport attribute batset 0/3/0 0/3/31 reverse-pole-pulse
enable
huawei(config-pstnport)#display pstnport attribute 0/3
  ---------------------------------------------------------------------
  F  /S  /P             0/3 /0
  ReversePolepulse        Enable
  PulseLevel             100(ms)
  PolarityReverseMode     Hard-polarity-reverse
  Dial-Mode              DTMF-Pulse-Both
  LineLock              Enable
  NlpMode               Nlp normal mode
  PolarityReverseWhenCLIP Disable
  PulsePeriodUpperLimit   200(ms)
  PulsePeriodLowerLimit   50(ms)
  PulseDurationUpperLimit 90(ms)
  PulseDurationLowerLimit 30(ms)
  PulsePauseUpperLimit    90(ms)
  PulsePauseLowerLimit    30(ms)
  OffhookTime(Idle)       80(ms)
  OffhookTime(Ring)       200(ms)
  OffhookTime(Fsk)        50(ms)
  ---------------------------------------------------------------------
  F  /S  /P             0/3 /1
  ReversePolepulse        Enable
  PulseLevel             100(ms)
  PolarityReverseMode     Hard-polarity-reverse
```

```
Dial-Mode            DTMF-Pulse-Both
LineLock             Enable
NlpMode              Nlp normal mode
PolarityReverseWhenCLIP Disable
PulsePeriodUpperLimit  200(ms)
PulsePeriodLowerLimit  50(ms)
PulseDurationUpperLimit 90(ms)
PulseDurationLowerLimit 30(ms)
PulsePauseUpperLimit   90(ms)
PulsePauseLowerLimit   30(ms)
OffhookTime(Idle)      80(ms)
OffhookTime(Ring)      200(ms)
OffhookTime(Fsk)       50(ms)
----------------------------------------------------------------------------
F  /S  /P            0/3 /31
ReversePolepulse       Enable
PulseLevel            100(ms)
PolarityReverseMode    Hard-polarity-reverse
Dial-Mode            DTMF-Pulse-Both
LineLock             Enable
NlpMode              Nlp normal mode
PolarityReverseWhenCLIP Disable
PulsePeriodUpperLimit  200(ms)
PulsePeriodLowerLimit  50(ms)
PulseDurationUpperLimit 90(ms)
PulseDurationLowerLimit 30(ms)
PulsePauseUpperLimit   90(ms)
PulsePauseLowerLimit   30(ms)
OffhookTime(Idle)      80(ms)
OffhookTime(Ring)      200(ms)
OffhookTime(Fsk)       50(ms)
----------------------------------------------------------------------------
```

NOTE

When a call begins and ends, the MG shows the start time and the end time based on the polarity reversal on the subscriber line. The billing terminal that supports the polarity reversal accounting function, such as a charging phone set, implements the polarity reversal accounting function based on the start time and the end time of a call.

1.18.2.6 (Optional) Configuring the Attributes of the Ringing Current

You can adjust the ringing volume and ringing tone by modifying the attributes of the ringing current. In general, the attributes of the ringing current do not need to be modified. You need to modify the attributes of the ringing current according to the local standards only when the default ringing current attributes do not meet the local standards.

Context

The attributes of the ringing current include the following two parameters:

- Ringing current frequency: A higher frequency indicates a sharper ringing tone. The default ringing current frequency is 25 Hz.
- AC amplitude (AC voltage): A greater amplitude indicates a louder ringing tone. The default AC amplitude is 75 Vrms.

Procedure

Step 1 In the global config mode, run the **voip** command to enter the VoIP mode.

Step 2 Run the **ring attribute set** command to configure the attributes of the ringing current according to the data plan.

Step 3 Run the **display ring attribute** command to check whether the attributes of the ringing current are the same as those in the data plan.

----End

Example

To set the ringing current frequency to 50 Hz (parameter value 2), AC amplitude to 50 Vrms (parameter value 2), do as follows:

```
huawei(config-voip)#ring attribute set frequency 2 acamplitude 2
huawei(config-voip)#display ring attribute
  ringing current frequency  : 50HZ
  ringing current acamplitute: 50VRMS
```

1.18.3 (Optional) Configuring Line Hunting

When multiple E1 lines exist in the upstream direction of a private branch exchange (PBX), hunting is performed based on the pilot number or other accounts required by the PBX. The line hunting function allows one or multiple accounts to share a group of ports by configuring hunting groups and hunting policies.

Context

- A hunting group consists of ports, subhunting groups, and hunting rules. A sub hunting group also consists of ports, sub hunting groups, and hunting rules.
- A wildcard number can be configured for hunting groups, for example, 024545*. You can also configure a direct dialing number.
- A port can belong to multiple hunting groups which must be in the same VAG.
- Only the Session Initiation Protocol (SIP) supports the line hunting function.

Procedure

Step 1 Run the **hunting-group add** command to add a hunting group. Then the hunting group is added to the specified SIP interface.

- **hunting-mode** indicates the hunting policy. This parameter can be set to **order**, **round-robin**, or **weighted-round-robin**.
  - **order**: indicates the sequential hunting.
  - **round-robin**: indicates the circular hunting.
  - **weighted-round-robin**: indicates the circular hunting by weight.

Step 2 Configure users on the SIP interface. Run different commands when configuring different users. Specifically,

- Run the **sippstnuser add** command when configuring PSTN users.
- Run the **sipbrauser add** command when configuring BRA users.

- Run the **sipprauser add** command when configuring PRA users.

Step 3 Run the **hunting-group member add** command to add hunting group members. A hunting group member can be a single port or a sub hunting group. After hunting group members are added, hunting is performed based on configurations.

Step 4 Run the **group-number add** command to add a hunting group account. After the group account is used by the hunting group, the account is called in based on hunting policies.

----End

Example

For example, when number 2878000 is dialed, hunting is cyclically performed between 0/2/1, 0/3/1, and 0/4/1 ports. When number 2878001 is dialed, the 0/2/1 is selected with preference. When the 0/2/1 port is busy, the 0/3/1 port is selected and then the 0/4/1 port. Configurations are as follows:

Parameter	HG1	HG2
hunting-mode	round-robin	order
inherit-flag	disable	disable
rotary	disable	disable
Group members	0/2/1, 0/3/1, 0/4/1	0/2/1, 0/3/1, 0/4/1

```
huawei(config)#interface sip 0
Are you sure to add the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#hunting-group add HG1 hunting-mode round-robin inherit-flag
disable
 rotary disable
huawei(config-if-sip-0)#hunting-group add HG2 hunting-mode order inherit-flag disable
rot
ary disable
huawei(config-if-sip-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser add 0/2/1 0
huawei(config-esl-user)#sippstnuser add 0/3/1 0
huawei(config-esl-user)#sippstnuser add 0/4/1 0
huawei(config-esl-user)#quit
huawei(config)#interface sip 0
huawei(config-if-sip-0)#hunting-group member add HG1 0/2/1 6
huawei(config-if-sip-0)#hunting-group member add HG1 0/3/1 7
huawei(config-if-sip-0)#hunting-group member add HG1 0/4/1 8
huawei(config-if-sip-0)#hunting-group member add HG2 0/2/1 6
huawei(config-if-sip-0)#hunting-group member add HG2 0/3/1 7
huawei(config-if-sip-0)#hunting-group member add HG2 0/4/1 8
huawei(config-if-sip-0)#group-number add 2878000 hunting-group HG1
huawei(config-if-sip-0)#group-number add 2878001 hunting-group HG2
```

1.19 Configuring the VoIP ISDN BRA Service (SIP-based)

After configuring the SIP interface, you can add VoIP ISDN BRA users on this interface to implement the VoIP ISDN BRA service.

Prerequisites

According to the actual network, a route from the MA5600T/MA5603T/MA5608T to the IMS must be configured to ensure that the MA5600T/MA5603T/MA5608T communicates with the IMS normally.

Context

- The ISDN is integrated services digital network. Defined by the International Telegraph and Telephone Consultative Committee (CCITT), the ISDN technology provides two types of user-network interfaces based on the user-network interface reference model.
  - ISDN basic rate interface (BRI), which supports a rate of 144 kbit/s and provides two B channels (for carrying services) and one D channel (for transmitting call control signaling and maintenance and management signaling). The rate of B channels is 64 kbit/s and the rate of D channel is 16 kbit/s. The service carried on the ISDN BRI is the ISDN basic rate access (BRA) service.
  - ISDN primary rate interface (PRI), which supports a rate of 2.048 Mbit/s and provides 30 B channels and one D channel. The rates of the B channel and D channel are both 64 kbit/s. The service carried on the ISDN PRI is the ISDN primary rate access (PRA) service.

The MA5600T/MA5603T/MA5608T can also function as a voice over IP gateway (VGW) in the IMS architecture. In the downstream direction, it is connected to the ISDN BRA users; in the upstream direction, it is connected to the IMS system through the SIP interface, providing the VoIP ISDN BRA service by working with the IMS core.

The functions and applications of the SIP interface are the same as the functions and applications of the MG interface.

Data preparation

Table 1-27 provides the data plan for configuring the VoIP ISDN BRA service.

Table 1-27 Data plan for configuring the VoIP ISDN BRA service when the SIP protocol is used

Item			Remarks
SIP interface data	Media and signaling parameters	Media and signaling upstream VLAN	It is used for the upstream VLAN of the VoIP service to be configured. **NOTICE** Note that the media and the signaling can use the same VLAN or different VLANs, depending on the negotiation with the upstream device.
		Signaling upstream port	Uplink port for configuring the SIP signaling.
		Media IP address and signaling IP	These IP addresses must be selected from the media and signaling IP

Item			Remarks
		address	address pools. The media and signaling IP address pools consists of all the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
		Default IP address of the MG	Next hop address from the MA5600T/MA5603T/MA5608T to the IMS core network device. **NOTICE** If the media IP address and the signaling IP address are different and the media and the signaling are transmitted upstream through different gateways, ensure that they correspond to correct gateways. Otherwise, the call service may fail.
	Attributes of the SIP interface **NOTE** Parameters listed here are mandatory, which means that the SIP interface cannot be enabled if these parameters are not configured.	SIP interface ID	It is used for the VoIP service to be configured.
		Signaling port ID of the SIP interface	The value range is 5000-5999. The protocol defines the port ID as 5060.
		IP address of the active IMS core network device to which the SIP interface belongs	When dual homing is not configured, parameters of only one IMS core network device are required. When dual homing is required, the IP address and the port ID of the standby IMS core network device must be configured.
		Port ID of the active IMS core network device to which the SIP interface belongs	
		Transmission mode of the SIP interface	The transmission mode is selected according to the requirements of the IMS core network device. Generally, UDP is used.
		Home domain of the SIP interface	It corresponds to parameter **home-domain** in the MG interface attributes.
		Index of the profile used by the SIP interface	It corresponds to parameter **Profile-index** in the MG interface attributes.
		IP address obtaining mode of the proxy server	• In the IP mode, the IP address and the port ID of the active proxy server must be configured. • In the DNS-A or DNS-SRV mode, the domain name of the active

Item			Remarks
			proxy server must be configured.
ISDN BRA user data (The data configuration must be consistent with the data configuration on the IMS.)	Slot that houses the BRA service board.		-
	User data	Phone number	The phone number that the IMS core network device allocates to the user must be configured.
		User priority	According to the service requirements, user priorities must be specified, including: - cat1: government1 (category 1 government user) - cat2: government2 (category 2 government user) - cat3: common (default priority, namely, common users)
	System parameters		The system parameters including the oversea version flag and message waiting indication (MWI) mode need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	Overseas parameters		The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	Attributes of an ISDN BRA Port		The attributes such as the working mode, remote power supply status, and auto-deactivation status of the port can be configured. Modify such attributes only if there is a special requirement.

1.19.1 Configuring the SIP Interface

As an interface used for the intercommunication between the MA5600T/MA5603T/MA5608T and the MGC, the interface is vital to the SIP-based VoIP service. Therefore, to implement the VoIP service, the SIP interface must be configured and must be in the normal state.

Procedure

1.19.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.19.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.19.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

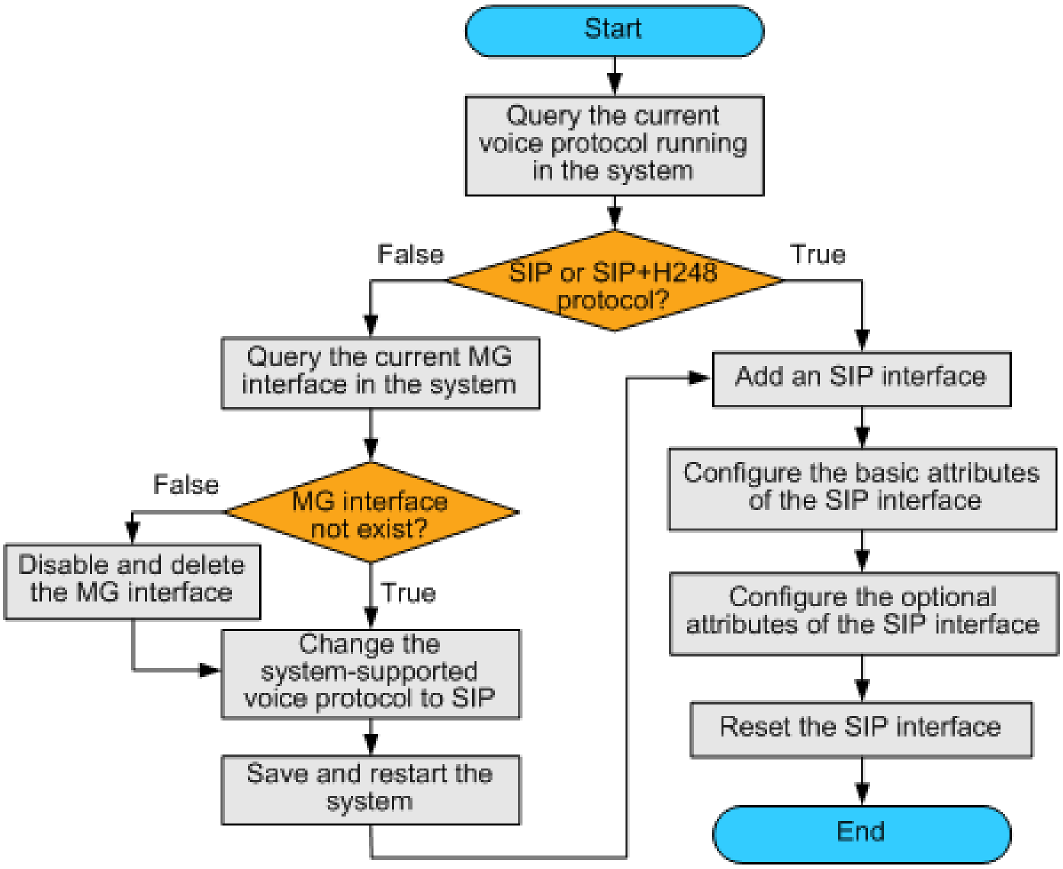
1.19.1.3 Adding an SIP Interface

The MA5600T/MA5603T/MA5608T exchanges the signaling and protocol packets with the information management system (IMS) through an SIP interface. To ensure uninterrupted signaling and protocol packet exchange, ensure that the status of the SIP interface is in a normal state after you add it.

Context

- One MA5600T/MA5603T/MA5608T supports up to eight SIP interfaces. Each SIP interface can be configured with the interface attributes separately.
- The SIP attributes configured for an SIP interface take effect on this interface only.
- The attributes of an SIP interface, including the signaling IP address, media IP address, transport-layer protocol, port ID of the transport-layer protocol, IP address (or domain name) of the proxy server, port ID of the proxy server, home domain name, profile index, are mandatory. In addition, the attributes of an SIP interface must be consistent with those configured on the IMS side so that the status of the SIP interface is normal.

Configuration Flowchart


Start
Query the current voice protocol running in the system
SIP or SIP+H248 protocol?
False
True
Query the current MG interface in the system
MG interface not exist?
False
True
Disable and delete the MG interface
Change the system-supported voice protocol to SIP
Save and restart the system
Add an SIP interface
Configure the basic attributes of the SIP interface
Configure the optional attributes of the SIP interface
Reset the SIP interface
End

Procedure

Step 1 Query the current voice protocol running in the system.

Run the **display protocol support** command to query the voice protocol that is currently supported by the system.

- If the system voice protocol is the SIP protocol, go to Step 6.
- If the system voice protocol is not the SIP protocol, go to Step 2.

Step 2 Query the current MG interface in the system.

Run the **display if-mgcp all** or **display if-h248 all** command to query whether an MGCP interface or an H.248 interface currently exists in the system.

- If there is no such an MG interface, go to Step 4.
- If there is such an MG interface, go to Step 3.

Step 3 Disable and delete the MG interface.

1. Delete the user data of this MG interface, and then run the **shutdown(mgcp)** or **shutdown(h248)** command to disable the MG interface according to the protocol type of the interface.

NOTICE

This operation interrupts all the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation. Before performing this operation, you must check whether the operation is allowed.

2. Run the **undo interface mgcp** or **undo interface h248** command to delete the MG interface.

Step 4 Change the system-supported voice protocol to SIP.

Run the **protocol support** command to change the system-supported voice protocol to SIP.

Step 5 Save the configuration data and restart the system.

Save the configuration data by running the **save** command. Then run the **reboot** command to restart the system to make the new configuration data take effect.

Step 6 Run the **interface sip** command to add an SIP interface.

Step 7 Run the **if-sip attribute basic** command to configure the basic attributes of the SIP interface.

NOTE

- Between **proxy IP** and **proxy domain**, which are basic attributes, at least one attribute must be configured. If both attributes are configured, the system determines which attribute is used according to the configured address mode of the proxy server.
- The profile index must be configured.

Step 8 Run the **if-sip attribute optional** command to configure the optional attributes of the SIP interface.

The optional attributes include some new service types supported by the SIP interface, such as the conference call, message waiting indicator, UA-profile subscription, and REG-state subscription, and also include the SIP proxy configuration mode. The optional attributes require the IMS-side support, and must be consistent with those on the IMA-side.

After the configuration is completed, run the **display if-sip attribute config** command to query the attributes of the SIP interface.

Step 9 Reset the SIP interface.

Run the **reset** command to reset the SIP interface for the new configuration data to take effect. Otherwise, the configuration data does not take effect but is only stored in the database.

After the SIP interface is reset successfully, run the **display if-sip attribute running** command to query the running status of the SIP interface. If the active (or standby) proxy is **up**, the SIP interface is normal.

----End

Example

Assume that: the media IP address is 10.10.10.13, signaling IP address is 10.10.10.13, transmission protocol is UDP, port ID is 5000, active proxy server IP address 1 is 10.10.10.14, port ID of the active proxy server is 5060, active proxy server domain name is proxy.domain, standby proxy server IP address 1 is 10.10.10.15, port ID of the standby proxy server is 5060, home domain name is sip.huawei.com, and profile index is 1. To configure the attributes of SIP interface 0, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#if-sip attribute basic media-ip 10.10.10.13 signal-ip 10
.10.10.13 signal-port 5000 transfer udp primary-proxy-ip1 10.10.10.14 primary-pr
oxy-port 5060 primary-proxy-domain proxy.domain secondary-proxy-ip1 10.10.10.15
secondary-proxy-port 5060 home-domain sip.huawei.com sipprofile-index 1
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#
  Resetting SIP interface 0 succeeded
huawei(config-if-sip-0)#display if-sip attribute running
  ------------------------------------------------------------
  ...//The rest information in response to this command is omitted.
  Primary Proxy State                up   //Indicates that the SIP interface is in the
normal state.
  Secondary Proxy State              down
  ...
  ------------------------------------------------------------
```

1.19.1.4 (Optional)Configuring the Software Parameters of an SIP Interface

The software parameters of a SIP interface mainly define certain common service attributes of the SIP interface. After the software parameter configuration, the parameters take effect immediately and are valid only to the SIP interface. Skip this topic if the system default configuration meets user requirements.

Prerequisites

The MG interface has been configured. For details about how to configure the MG interface, see 1.19.1 Configuring the SIP Interface.

Context

The details about the software parameters that can be configured of a SIP interface that supports SIP, see section **Usage Guidelines** in **mg-software parameter**. The other parameters are reserved in the system.

Table 1-28 lists parameters that are usually configured to a non-default value. The other parameters are not required.

Table 1-28 Software parameters usually configured of a SIP interface

Parameter	Description	Default Setting
2	Indicates whether the standalone mode is supported.	Numeral type. Range: 0-1. • 0: indicates that the standalone function is not supported. • 1: indicates that the standalone function is supported. Default: 0 This parameter is usually set to **1**.

Parameter	Description	Default Setting
8	Indicates whether the heartbeat message of the MG is disabled.	Numeral type. Range: 0-1. • 0: the heartbeat message of the MG is disabled • 1: the heartbeat message of the MG is enabled Default value: 0.

Procedure

Step 1 Enter the SIP interface mode.

In global config mode, run the **interface sip** command to enter the SIP interface mode.

Step 2 Configure software parameters.

Run the **mg-software parameter** command the software parameters required in the data plan.

Step 3 Check whether the software parameters are the same as those in the data plan.

Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 2 of the SIP interface 0 to 1 so that the SIP interface supports standalone, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#mg-software parameter 2 1
huawei(config-if-sip-0)#display mg-software parameter 2
  -------------------------------------------------
  MGID:0         para index:2   value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
 Parameter Index:  Interface software parameter name:
   2 : SAL Support
       0: No
       1: Yes
```

1.19.2 Configuring the VoIP ISDN BRA User

This topic describes how to configure the VoIP ISDN BRA user. After the SIP interface is configured, you can add the VoIP ISDN BRA user on this interface to implement the VoIP ISDN BRA service.

1.19.2.1 Configuring the ISDN BRA User Data

This topic describes how to configure the ISDN BRA user data on the SIP interface (the data must be the same as the corresponding data on the IMS) so that the ISDN BRA user can access the network to use the ISDN BRA service.

Prerequisites

The ISDN BRA service board must be inserted into the planned slot correctly and the board must be in the normal state.

Default Configuration

Table 1-29 lists the default settings of the attributes of the ISDN BRA user. When configuring the attributes of these attributes, you can modify the values according to the service requirements.

Table 1-29 Default settings of the attributes of the ISDN BRA user

Parameter	Default Settings
Priority of the ISDN BRA user	cat3 (common users)
Flag of reporting the UNI fault of the ISDN BRA user	disable
Threshold for the number of auto recoveries from deterioration faults	20
The matching scheme of an outgoing calling number	match
The voice bear capability of a port	speech+3.1k-audio
The type of the called number sent to the ISDN terminal when an incoming call is initiated	unknown
The international prefix flag	disable
The national prefix flag	disable
The change plan of an incoming called number used by the user on the port	-
The change plan of an outgoing calling number used by the user on the port	-
The calling number	-
The digitmap group used by the user	-

Procedure

Step 1 In global config mode, run the **esl user** command to enter the ESL user mode.

Step 2 Run the **sipbrauser add** command to add an ISDN BRA user.

Step 3 Run the **display sipbrauser** command to check whether the ISDN BRA user data is the same as the data plan.

Step 4 (Perform this step when you need to modify the attributes of an ISDN BRA user.) Run the **sipbrauser attribute set** command to configure the attributes of the ISDN BRA user.

Step 5 (Perform this step only after you modify the attributes of the ISDN BRA user.) Run the **display sipbrauser attribute** command to query whether the configured attributes of the ISDN BRA user are the same as the data plan.

Step 6 (Perform this step only when you need to configure an extended phone number for the ISDN BRA user.) Run the **sipbrauser extend-telno add** command to add multiple phone numbers or a phone number containing non-digit characters for the ISDN BRA user.

Step 7 (Perform this step only after you configure the extended phone number for an ISDN BRA user.) Run the **display sipbrauser extend-telno** command to query whether the configured extended phone number of the ISDN BRA user is the same as the data plan.

----End

Example

Assume that:

- SIP interface ID: 0
- Phone number: 28780000
- Call priority: cat3
- UNI fault report flag: disable
- Number of auto recoveries from deterioration faults: 10
- Voice bear capability: speech
- Type of the called number: national
- International prefix flag: enable
- national prefix flag: enable
- Digitmap group: DigitmapGroup1
- Change plan of an incoming called number: NumberChange1
- Change plan of an outgoing calling number: NumberChange2
- Calling number is 12345678
- Extended phone number: +86-755-28780000

To add such an ISDN BRA user connected to the 0/4/0 port, do as follows:

```
huawei(config)#esl-user
huawei(config-esl-user)#sipbrauser add 0/4/0 0 telno 28780000
huawei(config-esl-user)#display sipbrauser 0/4/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
         display sipbrauser 0/4/0
  ----------------------------------------
  F  /S /P    MGID     TelNo
  ----------------------------------------
  0/4/0    0       28780000
  ---------------------------------------
huawei(config-esl-user)#sipbrauser attribute set 0/4/0 priority cat3 unireport d
isable auto-resume-limit 10 bc speech called-num-type national international-pre
fix-flag enable national-prefix-flag enable digitmap-group DigitmapGroup1 incomi
ngcall-numberchange NumberChange1 outgoingcall-numberchange NumberChange2 cli-nu
mber 12345678
huawei(config-esl-user)#display sipbrauser attribute 0/4/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
         display sipbrauser attribute 0/4/0
  ----------------------------------------------------------------------------
  F  /S /P                       : 0/4/0
  UNIreport                      : disable
  Priority                       : cat3
  Auto reservice times/limit        : 0/10
  Digitmap group                  : DigitmapGroup1
  Incoming called number change plan : NumberChange1
  Outgoing caller number change plan : NumberChange2
  CLI number                     : 12345678
  CLI mode                       : match
  Bear capability                 : speech
  Called number type              : national
  International prefix flag        : enable
  National prefix flag            : enable
  DSP-para-template               : -
  ----------------------------------------------------------------------------
huawei(config-esl-user)#sipbrauser extend-telno add 0/4/0 +86-755-28780000
huawei(config-esl-user)#display sipbrauser extend-telno 0/4/0
  -----------------------------------------------------------
     Index          Extend-telno
   -----------------------------------------------------------
     1             +86-755-28780000
     2             28780000
     3             0755-28780000
   -----------------------------------------------------------
```

1.19.2.2 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  -----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
  -----------------------------------------------------------------------------
```

1.19.2.3 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1

  Command:
        display oversea parameters 1
    -----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 100
  Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
  -----------------------------------------------------------------------------
```

1.19.2.4 (Optional) Configuring the Attributes of the ISDN BRA Port

This topic describes how to configure the attributes of an ISDN BRA port to ensure that the ISDN BRA port can meet the actual application requirements. You can configure the auto-deactivation status, remote power supply status, UNI fault alarming function, and working mode of the port.

Default Configuration

Table 1-30 lists the default values of the attributes of an ISDN BRA port. When configuring these attributes, you can modify the values according to the service requirements.

Table 1-30 Default values of the attributes of an ISDN BRA port

Parameter	Default Setting
Autodeactive	Disable
Autodeactive-delay	30s
Activemode	unstable-active
Remotepower	Disable
Unialarm	Disable
Workmode	p2mp

Procedure

Step 1 In global config mode, run the **braport** command to enter braport mode.

Step 2 Run the **braport attribute set** command to configure the attributes such as the working mode, auto-deactivation status, and remote power supply status of the port.

If an ISDN BRA port needs to be connected to multiple terminal users, configure the working mode of the port to p2mp. If an ISDN BRA port needs to be connected to only one terminal user, configure the working mode of the port to p2p.

For detailed description of the **braport attribute set** command, see the parameter description in **braport attribute set**.

----End

Example

Assume that the working mode is p2mp, the activation mode is stable, and the auto-deactivation function is disabled. To configure such attributes of ISDN BRA port 0/4/0, do as follows:

```
huawei(config)#braport
huawei(config-braport)#braport attribute set 0/4/0 workmode p2mp activemode
stable-active
huawei(config-braport)#display braport attribute
{ frameid/slotid/portid<S><Length 1-15>|frameid/slotid<S><Length 1-15> }:0/4/0

  Command:
        display braport attribute 0/4/0
-----------------------------------------------------------------------------
F  /S /P  Remotepower Workmode Autodeactive Deactivedelay Activemode Unialarm
-----------------------------------------------------------------------------
0/4/0  disable     p2mp    disable     30          stable    disable
-----------------------------------------------------------------------------
```

1.20 Configuring the VoIP ISDN PRA Service (SIP-based)

After configuring the SIP interface, you can add VoIP ISDN users on this interface to implement the VoIP ISDN service.

Prerequisites

According to the actual network, a route from the Access node to the IMS must be configured to ensure that the Access node communicates with the IMS normally.

Context

- The voice over IP (VoIP) service uses the IP packet switched network for transmission after the traditional analog voice signals are compressed and packetized. This service lowers the cost of the voice service. For the detailed description of the VoIP service, see Voice Feature in the *Feature Description*.
- Defined by the International Telegraph and Telephone Consultative Committee (CCITT), the ISDN is a communication network evolved from the Integrated Digital Network (IDN). The ISDN service provides the E2E digital connection and supports multiple types of voice and non-voice telecom services. On the ISDN network, users can access the network through the following two interfaces: (The ISDN technology provides two types of user-network interfaces based on the user-network interface reference model.)
  - ISDN basic rate interface (BRI), which supports a rate of 144 kbit/s and provides two B channels (for carrying services) and one D channel (for transmitting call control signaling and maintenance and management signaling). The rate of B channels is 64 kbit/s and the rate of D channel is 16 kbit/s. The service carried on the ISDN BRI is the ISDN basic rate access (BRA) service.
  - ISDN primary rate interface (PRI), which supports a rate of 2.048 Mbit/s and provides 30 B channels and one D channel. The rates of the B channel and D channel are both 64 kbit/s. The service carried on the ISDN PRI is the ISDN primary rate access (PRA) service.

The Access node can also function as a voice over IP gateway (VGW) in the IMS architecture. In the downstream direction, it is connected to the ISDN PRA users; in the upstream direction, it is connected to the IMS system through the SIP interface, providing the VoIP ISDN PRA service by working with the IMS core.

The functions and applications of the SIP interface are the same as the functions and applications of the MG interface.

Data preparation

Table 1-31 provides the data plan for configuring the VoIP ISDN PRA service.

Table 1-31 Data plan for configuring the VoIP ISDN PRA service when the SIP protocol is used

Item			Remarks
SIP interface data	Media and signaling parameters	Media and signaling upstream VLAN	It is used for the upstream VLAN of the VoIP service to be configured. **NOTICE** Note that the media and the signaling can use the same VLAN or different VLANs, depending on the negotiation with the

Item			Remarks
			upstream device.
		Signaling upstream port	Uplink port for configuring the SIP signaling.
		Media IP address and signaling IP address	These IP addresses must be selected from the media and signaling IP address pools. The media and signaling IP address pools consists of all the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
		Default IP address of the MG	Next hop address from the Access node to the IMS core network device. **NOTICE** If the media IP address and the signaling IP address are different and the media and the signaling are transmitted upstream through different gateways, ensure that they correspond to correct gateways. Otherwise, the call service may fail.
	Attributes of the SIP interface **NOTE** Parameters listed here are mandatory, which means that the SIP interface cannot be enabled if these parameters are not configured.	SIP interface ID	It is used for the VoIP service to be configured.
		Signaling port ID of the SIP interface	The value range is 5000-5999. The protocol defines the port ID as 5060.
		IP address of the active IMS core network device to which the SIP interface belongs	When dual homing is not configured, parameters of only one IMS core network device are required. When dual homing is required, the IP address and the port ID of the standby IMS core network device must be configured.
		Port ID of the active IMS core network device to which the SIP interface belongs	
		Transmission mode of the SIP interface	The transmission mode is selected according to the requirements of the IMS core network device. Generally, UDP is used.
		Home domain of the SIP interface	It corresponds to parameter **home-domain** in the MG interface attributes.
		Index of the profile used by the SIP interface	It corresponds to parameter **Profile-index** in the MG interface attributes.
		IP address obtaining mode	

Item			Remarks
		of the proxy server	• In the IP mode, the IP address and the port ID of the active proxy server must be configured. • In the DNS-A or DNS-SRV mode, the domain name of the active proxy server must be configured.
ISDN PRA user data (The data configuration must be consistent with the data configuration on the IMS.)	Slot that houses the E1 service board.		-
	User data	Phone number	The phone number that the IMS core network device allocates to the user must be configured.
		User priority	According to the service requirements, user priorities must be specified, including: • cat1: government1 (category 1 government user) • cat2: government2 (category 2 government user) • cat3: common (default priority, namely, common users)
	Centrex		The out-centrex prefix and out-centrex attributes of a centrex need to be configured according to local standards.
	System parameters		The system parameters including the oversea version flag and message waiting indication (MWI) mode need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	Overseas parameters		The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	E1 Port attributes		Board access mode, port mode, line coding mode, and port impedance need to be configured, and need not be modified if there is no special requirement.

1.20.1 Configuring the SIP Interface

As an interface used for the intercommunication between the MA5600T/MA5603T/MA5608T and the MGC, the interface is vital to the SIP-based VoIP service. Therefore, to implement the VoIP service, the SIP interface must be configured and must be in the normal state.

Procedure

1.20.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
```

```
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.20.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.20.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.

2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

1.20.1.3 Adding an SIP Interface

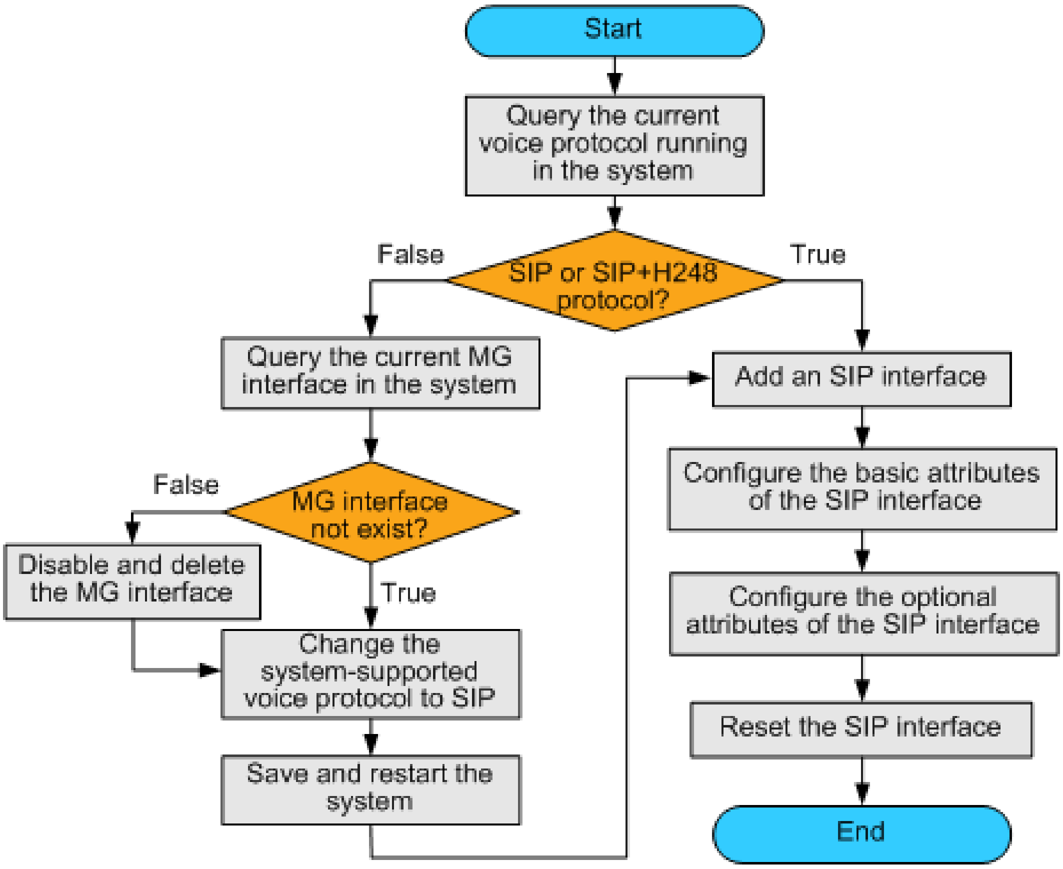
The MA5600T/MA5603T/MA5608T exchanges the signaling and protocol packets with the information management system (IMS) through an SIP interface. To ensure uninterrupted signaling and protocol packet exchange, ensure that the status of the SIP interface is in a normal state after you add it.

Context

- One MA5600T/MA5603T/MA5608T supports up to eight SIP interfaces. Each SIP interface can be configured with the interface attributes separately.
- The SIP attributes configured for an SIP interface take effect on this interface only.

- The attributes of an SIP interface, including the signaling IP address, media IP address, transport-layer protocol, port ID of the transport-layer protocol, IP address (or domain name) of the proxy server, port ID of the proxy server, home domain name, profile index, are mandatory. In addition, the attributes of an SIP interface must be consistent with those configured on the IMS side so that the status of the SIP interface is normal.

Configuration Flowchart


Start
Query the current voice protocol running in the system
SIP or SIP+H248 protocol?
False
True
Query the current MG interface in the system
MG interface not exist?
False
True
Disable and delete the MG interface
Change the system-supported voice protocol to SIP
Save and restart the system
Add an SIP interface
Configure the basic attributes of the SIP interface
Configure the optional attributes of the SIP interface
Reset the SIP interface
End

Procedure

Step 1 Query the current voice protocol running in the system.

Run the **display protocol support** command to query the voice protocol that is currently supported by the system.

- If the system voice protocol is the SIP protocol, go to Step 6.
- If the system voice protocol is not the SIP protocol, go to Step 2.

Step 2 Query the current MG interface in the system.

Run the **display if-mgcp all** or **display if-h248 all** command to query whether an MGCP interface or an H.248 interface currently exists in the system.

- If there is no such an MG interface, go to Step 4.
- If there is such an MG interface, go to Step 3.

Step 3 Disable and delete the MG interface.

1. Delete the user data of this MG interface, and then run the **shutdown(mgcp)** or **shutdown(h248)** command to disable the MG interface according to the protocol type of the interface.

NOTICE

This operation interrupts all the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation. Before performing this operation, you must check whether the operation is allowed.

2. Run the **undo interface mgcp** or **undo interface h248** command to delete the MG interface.

Step 4 Change the system-supported voice protocol to SIP.

Run the **protocol support** command to change the system-supported voice protocol to SIP.

Step 5 Save the configuration data and restart the system.

Save the configuration data by running the **save** command. Then run the **reboot** command to restart the system to make the new configuration data take effect.

Step 6 Run the **interface sip** command to add an SIP interface.

Step 7 Run the **if-sip attribute basic** command to configure the basic attributes of the SIP interface.

NOTE

- Between **proxy IP** and **proxy domain**, which are basic attributes, at least one attribute must be configured. If both attributes are configured, the system determines which attribute is used according to the configured address mode of the proxy server.
- The profile index must be configured.

Step 8 Run the **if-sip attribute optional** command to configure the optional attributes of the SIP interface.

The optional attributes include some new service types supported by the SIP interface, such as the conference call, message waiting indicator, UA-profile subscription, and REG-state subscription, and also include the SIP proxy configuration mode. The optional attributes require the IMS-side support, and must be consistent with those on the IMA-side.

After the configuration is completed, run the **display if-sip attribute config** command to query the attributes of the SIP interface.

Step 9 Reset the SIP interface.

Run the **reset** command to reset the SIP interface for the new configuration data to take effect. Otherwise, the configuration data does not take effect but is only stored in the database.

After the SIP interface is reset successfully, run the **display if-sip attribute running** command to query the running status of the SIP interface. If the active (or standby) proxy is **up**, the SIP interface is normal.

----End

Example

Assume that: the media IP address is 10.10.10.13, signaling IP address is 10.10.10.13, transmission protocol is UDP, port ID is 5000, active proxy server IP address 1 is 10.10.10.14, port ID of the active proxy server is 5060, active proxy server domain name is proxy.domain, standby proxy server IP address 1 is 10.10.10.15, port ID of the standby proxy server is 5060, home domain name is sip.huawei.com, and profile index is 1. To configure the attributes of SIP interface 0, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#if-sip attribute basic media-ip 10.10.10.13 signal-ip 10.10.10.13 signal-port 5000 transfer udp primary-proxy-ip1 10.10.10.14 primary-proxy-port 5060 primary-proxy-domain proxy.domain secondary-proxy-ip1 10.10.10.15 secondary-proxy-port 5060 home-domain sip.huawei.com sipprofile-index 1
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#
  Resetting SIP interface 0 succeeded
huawei(config-if-sip-0)#display if-sip attribute running
  ------------------------------------------------------------
  ...//The rest information in response to this command is omitted.
  Primary Proxy State                up   //Indicates that the SIP interface is in the normal state.
  Secondary Proxy State              down
  ...
  ------------------------------------------------------------
```

1.20.1.4 (Optional)Configuring the Software Parameters of an SIP Interface

The software parameters of a SIP interface mainly define certain common service attributes of the SIP interface. After the software parameter configuration, the parameters take effect immediately and are valid only to the SIP interface. Skip this topic if the system default configuration meets user requirements.

Prerequisites

The MG interface has been configured. For details about how to configure the MG interface, see 1.20.1 Configuring the SIP Interface.

Context

The details about the software parameters that can be configured of a SIP interface that supports SIP, see section **Usage Guidelines** in **mg-software parameter**. The other parameters are reserved in the system.

Table 1-32 lists parameters that are usually configured to a non-default value. The other parameters are not required.

Table 1-32 Software parameters usually configured of a SIP interface

Parameter	Description	Default Setting
2	Indicates whether the standalone mode is supported.	Numeral type. Range: 0-1. • 0: indicates that the standalone function is not supported. • 1: indicates that the standalone function is supported.

Parameter	Description	Default Setting
		Default: 0 This parameter is usually set to **1**.
8	Indicates whether the heartbeat message of the MG is disabled.	Numeral type. Range: 0-1. • 0: the heartbeat message of the MG is disabled • 1: the heartbeat message of the MG is enabled Default value: 0.

Procedure

Step 1 Enter the SIP interface mode.

In global config mode, run the **interface sip** command to enter the SIP interface mode.

Step 2 Configure software parameters.

Run the **mg-software parameter** command the software parameters required in the data plan.

Step 3 Check whether the software parameters are the same as those in the data plan.

Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 2 of the SIP interface 0 to 1 so that the SIP interface supports standalone, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#mg-software parameter 2 1
huawei(config-if-sip-0)#display mg-software parameter 2
  ------------------------------------------------
  MGID:0         para index:2   value:1
  ------------------------------------------------
 APPENDIX:
  ------------------------------------------------
 Parameter Index:  Interface software parameter name:
   2 : SAL Support
       0: No
       1: Yes
```

1.20.2 Configuring the VoIP ISDN PRA User

This topic describes how to configure the VoIP ISDN PRA user. After the MG interface is configured, you can add the VoIP ISDN PRA user on this interface to implement the VoIP ISDN PRA service.

1.20.2.1 Configuring the Attributes of the E1 Port

This topic describes how to configure the attributes of the E1 port to ensure that the ISDN PRA port meets the actual application requirements.

Context

You can configure the impedance, line coding mode, and working mode of the E1 port.

Default Configuration

Table 1-33 lists the default values of the E1 port. When configuring the attributes of the E1 port, you need to modify the values according to the service requirements.

Table 1-33 Default values of the E1 port

Parameter	Default Setting
Port impedance	E1 mode: 75 ohm
Line coding mode	E1 mode: HDB3
CRC4	Enable
The mode for digital section access	Digital
Signaling type	CCS

Procedure

Step 1 In global config mode, run the **interface edt** command to enter the EDT mode.

Step 2 (Optional) Run the **e1port impedance** command to configure the impedance of an E1 port.

Step 3 (Optional; perform this step when you need to modify the line coding mode of the port) Run the **e1port line-code** command to configure the line coding mode of the E1 port.

NOTE

In E1 mode, the system supports two line coding modes, namely, HDB3 and AMI.

Step 4 (Optional) Run the **e1port crc4** command to configure the CRC4 function of an E1 port.

Step 5 (Optional) Run the **e1port attribute set** command to configure the digital section access mode of an E1 port.

Step 6 (Optional) Run the **e1port signal** command to configure the signaling type of an E1 port.

----End

Example

Assume that the E1 ports on ISDN PRA board work in HDB3 line encoding mode, the CRC4 function is enable, To configure such E1 ports, do as follows:

```
huawei(config)#interface edt 0/1
huawei(config-if-edt-0/1)#e1port line-code 1 HDB3
huawei(config-if-edt-0/1)#display e1port line-code 1
  ---------------------
  F/S/P  linecode
  ---------------------
  0/1/1  HDB3
  ---------------------
huawei(config-if-edt-0/1)#e1port crc4 1 enable
huawei(config-if-edt-0/1)#display e1port attribute 1
  ----------------------------------------------------------
   F/S/P   Signaltype   CRC4    Impedance   Accessmode
  ----------------------------------------------------------
   0/1/1   CCS         Enable  75         Digital
  ----------------------------------------------------------
```

1.20.2.2 Configuring the ISDN PRA User Data

This topic describes how to configure the ISDN PRA user data on the SIP interface (the data must be the same as the corresponding data on the IMS) so that the ISDN PRA user can access the network to use the ISDN PRA service.

Prerequisites

The ISDN PRA service board must be inserted into the planned slot correctly and the board must be in the normal state.

Default Configuration

Table 1-34 lists the default settings of the attributes of the ISDN PRA user. When configuring the attributes of these attributes, you can modify the values according to the service requirements.

Table 1-34 Default settings of the attributes of the ISDN PRA user

Parameter	Default Settings
Priority of the ISDN PRA user	cat3 (common users)
Flag of reporting the UNI fault of the ISDN PRA user	disable
Sub-channel active mask of the ISDN PRA user	255.255.255.255
Threshold for the number of auto recoveries from deterioration faults	20
The matching scheme of an outgoing calling number	match
The voice bear capability of a port	speech+3.1k-audio
The type of the called number sent to the ISDN terminal when an incoming call is initiated	unknown
The international prefix flag	disable

Parameter	Default Settings
The national prefix flag	disable
The change plan of an incoming called number used by the user on the port	-
The change plan of an outgoing calling number used by the user on the port	-
The calling number	-
The digitmap group used by the user	-

Procedure

Step 1 In global config mode, run the **esl user** command to enter the ESL user mode.

Step 2 Run the **sipprauser add** command to add an ISDN PRA user.

Step 3 Run the **display sipprauser** command to check whether the ISDN PRA user data is the same as the data plan.

Step 4 (Perform this step when you need to modify the attributes of an ISDN PRA user.) Run the **sipprauser attribute set** command to configure the attributes of the ISDN PRA user.

Step 5 (Perform this step only after you modify the attributes of the ISDN PRA user.) Run the **display sipprauser attribute** command to query whether the configured attributes of the ISDN PRA user are the same as the data plan.

Step 6 (Perform this step only when you need to configure an extended phone number for the ISDN PRA user.) Run the **sipprauser extend-telno add** command to add multiple phone numbers or a phone number containing non-digit characters for the ISDN PRA user.

Step 7 (Perform this step only after you configure the extended phone number for an ISDN PRA user.) Run the **display sipprauser extend-telno** command to query whether the configured extended phone number of the ISDN PRA user is the same as the data plan.

----End

Example

Assume that:

- SIP interface ID: 0
- Phone number: 28780000
- Call priority: cat3
- UNI fault report flag: disable
- Number of auto recoveries from deterioration faults: 10
- Sub-channel active mask: 255.255.1.3
- Voice bear capability: speech
- Type of the called number: national
- International prefix flag: enable

- national prefix flag: enable
- Digitmap group: DigitmapGroup1
- Change plan of an incoming called number: NumberChange1
- Change plan of an outgoing calling number: NumberChange2
- Calling number is 12345678
- Extended phone number: +86-755-28780000

To add such an ISDN PRA user connected to the 0/1/0 port, do as follows:

```
huawei(config)#esl-user
huawei(config-esl-user)#sipprauser add 0/1/0 0 telno 28780000
huawei(config-esl-user)#display sipprauser 0/1/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
        display sipprauser 0/1/0
  ---------------------------------------
  F  /S /P   MGID    TelNo
  ---------------------------------------
  0/1/0   0      28780000
  ---------------------------------------
huawei(config-esl-user)#sipprauser attribute set 0/1/0 priority cat3 unireport d
isable auto-resume-limit 10 activemask 255.255.255.3 bc speech called-num-type n
ational international-prefix-flag enable national-prefix-flag enable digitmap-gr
oup DigitmapGroup1 incomingcall-numberchange NumberChange1 outgoingcall-numberch
ange NumberChange2 cli-number 12345678
huawei(config-esl-user)#display sipprauser attribute 0/1/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
        display sipprauser attribute 0/1/0
  ------------------------------------------------------------------------
  F  /S /P                      : 0/1/0
  UNIreport                      : disable
  Prior                         : cat3
  Mask of sub channel             : 255.255.255.3
  Auto reservice times/limit       : 0/10
  Digitmap group                  : DigitmapGroup1
  Incoming called number change plan : NumberChange1
  Outgoing caller number change plan : NumberChange2
  CLI number                     : 12345678
  CLI mode                      : match
  Bear capability                 : speech+3.1k-audio
  Called number type              : unknown
  International prefix flag        : disable
  National prefix flag            : disable
  ------------------------------------------------------------------------
huawei(config-esl-user)#sipprauser extend-telno add 0/1/0 +86-755-28780000
huawei(config-esl-user)#display sipprauser extend-telno 0/1/0
  -----------------------------------------------------------
    Index          Extend-telno
   -----------------------------------------------------------
    1             +86-755-28780000
    2             28780000
  3             0755-28780000
-----------------------------------------------------------
```

1.20.2.3 (Optional) Configuring the Centrex

Centrex refers to a virtual user group. The MA5600T/MA5603T/MA5608T supports the following functions: Members in a centrex can call each other by dialing short numbers, and members in a centrex can call the members outside of the centrex by dialing centrex prefix + the complete phone number. Generally, a softswitch issues centrex parameters. If a softswitch does not issue centrex parameters, use the parameters preset on the MA5600T/MA5603T/MA5608T. These parameter values must be the same as these on the softswitch.

Context

- Centrex prefix: When attempting to call a user in another centrex group, a user must dial the centrex prefix before dialing the called number. A centrex prefix contains 0 to 9 digits.
- The function that the members in a centrex can call each other by dialing short numbers need not be configured on the MA5600T/MA5603T/MA5608T through the command line interface (CLI).
- The function that the members in a centrex can call the members outside of the centrex by dialing centrex prefix + the complete phone number can be supported only when the SIP protocol is used.
- The centrex attribute of a centrex can be direct centrex or two-stage centrex. The similarity and difference are as follows:
  - Similarity: When the members in a centrex need to call the members outside of the centrex, they must dial the centrex prefix.
  - Difference: If the centrex attribute is set to two-stage centrex, the members in a centrex can hear the dial tone again after dialing the centrex prefix. If the centrex attribute is set to direct centrex, no out-group dial tone is played.

Procedure

Step 1 Configure the centrex call function for a centrex group.

The MA5600T/MA5603T/MA5608T supports the configuration of the centrex prefix through one of two methods. In method 1, configure the centrex prefix and centrex attributes for a single user in ESL user mode. In method 2, configure the centrex prefix and centrex attributes for all the users in global config mode. When both methods are used, method 1 takes effect.

- Use method 1:

1. In ESL user mode, run the **sippstnuser servicedata parameter set** command to configure the centrex prefix and centrex attributes of a centrex group.
2. Run the **display sippstnuser servicedata** command to check whether the centrex parameter settings of a centrex are the same as the data plan.

- Use method 2:

NOTE

If method 2 is used, the MA5600T/MA5603T/MA5608T uses the centrex digitmap to match the centrex prefix, and uses the call digitmap, or the normal digitmap, to match the phone number dialed by a user.

1. In global config mode, run the **local-digitmap add** command to configure a direct centrex digitmap or a two-stage centrex digitmap.

NOTE

- The system does not support the adding of a direct centrex digitmap and a two-stage centrex digitmap at the same time. Add a digitmap based on site requirements.
- When you add a direct centrex digitmap, the system centrex attribute is direct centrex. When you add a two-stage centrex digitmap, the system centrex attribute is two-stage centrex.

2. Run the **display local-digitmap** command to check whether the local digitmap is the same as the data plan.

Step 2 Check whether the value of the sipprofile control point 148 is the same as the data plan.

Run the **display sipprofile syspara detail** command to check whether the value of control point 148 is the same as the data plan. If they are different, run the **sipprofile modify** command in SIP mode to change the control point value.

NOTE

- The sipprofile control point 148 can be set to 0 or 1. When it is set to 0, a phone number does not contain a centrex prefix; when it is set to 1, a phone number contain a centrex prefix. By default, it is 1.
- Run the **if-sip attribute basic** *sipprofile-index 0* command to specify a user-defined profile for the current SIP interface, and run the **sipprofile modify** command to change the value for the sipprofile control point 148.

----End

MA5600T/MA5603T/MA5608T

Assume that the centrex prefix of the MA5600T/MA5603T/MA5608T user with phone number 88627792 is 8100, the centrex attribute is two-stage centrex, and the control point of the Sipprofile uses the default value.

To configure the centrex call function for such a user by using method 1, do as follows:

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser servicedata parameter set 0/2/1 telno 88627792
centrexprefix 8100 centrexflag dialsecondary
huawei(config-esl-user)#display sippstnuser servicedata 0/2/1 telno 88627792
  ----------------------------------------
  F /S /P               : 0/2/1
  telno                : 88627792
  centrexno             : -
  centrexprefix          : 8100
  centrexflag           : dialsecondary
  mwimode               : deferred
  hottime(s)            : 100
  hotlinenum            : -
  dialtone              : normal
  cfbnum               : -
  cfnrnum               : -
  cfunum               : -
  cfnrtime(s)           : 100
  displayname           : -
  permanent-hold-mode     : norecall
  permanent-hold-time(s)  : 20
  ---------------------------------------------------
```

To configure the centrex call function for such a user by using method 2, and plan the digitmap body to (8100) and the digitmap name to **huawei1** for the two-stage centrex digitmap according to the centrex prefix, do as follows:

```
huawei(config)#local-digitmap add huawei1 second-centrex (8100)
huawei(config)#display local-digitmap all
  -----------------------------------------
  Name    : huawei1
  Type    : second-centrex
  Body    : (8100)
  Protocol: sip
  -----------------------------------------
```

1.20.2.4 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  ----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
  ----------------------------------------------------------------------------
```

1.20.2.5 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1

  Command:
        display oversea parameters 1
    ----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 100
  Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
  ----------------------------------------------------------------------------
```

1.21 Configuring the VoIP PSTN Service (H.248-based or MGCP-based)

This topic describes how to configure the VoIP PSTN service when the protocol adopted by the Access node is H.248 or MGCP.

Application Context

The voice over IP (VoIP) service uses the IP packet switched network for transmission after the traditional analog voice signals are compressed and packetized, to lower the cost of the voice service.

In the NGN network, the Access node functions as an access gateway (AG) and exchanges signaling with the media gateway controller (MGC) through the MG control protocol (mainly H.248 and MGCP). In this way, the VoIP, FoIP, and MoIP services are implemented under the control of the MGC. The MG interface, as an interface for the communication between the Access node (AG) and the MGC, plays a decisive role in the H.248-based or MGCP-based VoIP service.

H.248, also called MeGaCo, is a protocol developed based on MGCP by combining the features of other media gateway control protocols. Compared with MGCP, H.248 supports more types of access technologies and supports mobility of terminals; however, the configuration of the H.248-based VoIP service is the same as that of the MGCP-based VoIP service.

Prerequisite

- To implement the VoIP service, the MG interface must be configured and must be in the normal state.
- According to the actual network, a route from the Access node to the MGC must be configured to ensure that the Access node and the MGC are reachable from each other.
- The voice daughter board on the control board works in the normal state.

- Electronic switch 1 must be in **location-0** (indicating that the VoIP service is supported) If the SCUB control board is used. For details about how to configure the electronic switch, see **electro-switch**.

Precaution

The MG control protocols (H.248 and MGCP) are master/slave protocols, and the MGC controls the AG to implement the call connection. Therefore, the data on the AG for interconnection with the MGC, including the attributes of the MG interface and the attributes of the VoIP user, must be the same as the corresponding data on the MGC. Before configuring the VoIP service, you must make the data plan by considering interconnection with the MGC.

Data preparation

Table 1-35 provides the data plan for configuring the VoIP service.

Table 1-35 Data plan for configuring the H.248-based or MGCP-based VoIP service

Item			Remarks
MG interface data (The data configuration must be the same as the data configuration on the MGC.)	Media and signaling parameters	Media and signaling upstream VLAN	It is used as the upstream VLAN of the VoIP service to be configured. Standard VLAN is recommended.
		Media and signaling upstream port	It is used as the upstream port of the VoIP service to be configured.
		Media and signaling IP addresses	These IP addresses must be selected from the media and signaling IP address pools. The media and signaling IP address pools consists of all the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN. **NOTICE** The MGCP interface on the Access node does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.
		Default IP address of the MG	It is the next hop IP address from the Access node to the MGC.
	Parameters of the MG interface **NOTE** Parameter	MG interface ID	It is the MG interface ID used for the VoIP service to be configured, which determines the virtual access gateway (VAG) specified for the user.
		Signaling port ID of the MG	It is the transport layer protocol port ID used for the signaling exchange

Item			Remarks
	s listed here are mandatory, which means that the MG interface fails to be enabled if these parameters are not configured.	interface	between the Access node (AG) and the MGC. • Default signaling port ID defined in H.248: 2944 (text) and 2945 (binary) • Default signaling port ID defined in MGCP: 2727
		IP address of the primary MGC to which the MG interface belongs	When dual homing is not configured, the parameters of the primary MGC need to be configured. When dual homing is configured, the IP address and the port ID of the secondary MGC also need to be configured.
		Port ID of the primary MGC to which the MG interface belongs	
		Coding mode of the MG interface	Currently, the **text** coding mode is supported. **NOTE** For the MG interface that supports MGCP, the default coding mode is the **text** coding mode. This parameter can be queried, but cannot be configured.
		Transmission mode of the MG interface	The transmission mode is selected according to the requirements on the MGC side. Generally, UDP is adopted. **NOTE** For the MG interface that supports MGCP, the default transmission mode is UDP. This parameter can be queried, but cannot be configured.
		Domain name of the MG interface	It corresponds to the parameter **domainName** of the MG interface. • When the MGCP protocol is used, this parameter must be configured. Otherwise, the MG interface fails to be enabled. • When the H.248 protocol is used, this parameter must be configured if the parameter **MIDType** of the H.248 message is configured as the domain name. Otherwise, the MG interface fails to be enabled. In other situations, this parameter is optional.
		Profile index of the MG interface	If the MGC interconnected with the MG is also made by Huawei, set the profile index to 1 or do not set it (it is

Item			Remarks
			1 by default); if the MGC interconnected with the MG is made by another manufacturer, set the profile index to the corresponding value of the manufacturer.
		Device name of the MG interface	It is supported by the H.248 protocol, and corresponds to the parameter **deviceName** of the MG interface that uses the H.248 protocol. When the H.248 protocol is used, this parameter must be configured if the parameter **MIDType** of the H.248 message is configured as the domain name. Otherwise, the MG interface fails to be enabled. In other situations, this parameter is optional.
	Digitmap of the MG interface		The digitmaps here are used for certain special applications such as emergency calls and emergency standalone. The digitmaps for common phone services are issued to the AG by the MGC, and therefore such digitmaps do not need to be configured on the AG. If the services such as emergency calls and emergency standalone are not required, the digitmaps do not need to be configured.
	Software parameters of the MG interface		According to the service requirements, the configuration of software parameters determines whether the MG interface supports the functions such as dual homing and emergency standalone.
	Ringing mode of the MG interface		According to the service requirements, the ringing modes of the MG interface need to be determined.
	Terminal ID (TID) format of the MG interface		The TID format determines the generation mode of various types of user terminals on an MG interface.
Voice user data (The data configuration must be the same as the data configuration	Slot of the voice service board		-
	User data	Phone number	The phone numbers allocated by the MGC need to be determined, and the paging numbers for users' emergency standalone need to be planned if the emergency standalone function is provided.

Item			Remarks
on the MGC.)		TID	If the TID template with which the PSTN user is bound does not support terminal layering, this parameter needs to be configured.
		User priority	According to the service requirements, user priority needs to be specified. The user priority includes the following: • cat1: government1 (category 1 government users) • cat2: government2 (category 2 government users) • cat3: common (common users)
		User type	According to the service requirements, user type needs to be specified. The user type includes the following: • DEL: direct exchange lines (default) • ECPBX: earth calling PBX • LCPBX: loop calling PBX • PayPhone: pay phone
	System parameters		The system parameters including the overseas version flag and message waiting indication (MWI) mode need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	Overseas parameters		The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.
	PSTN port attributes		If the PSTN port needs to support the polarity reversal accounting, the PSTN port needs to be configured to support the polarity reversal pulse. Other attributes do not need to be modified if there is no special requirement.
	Ringing current attributes		You can adjust the ringing volume by modifying the attributes of the ringing current. In general, the attributes of the ringing current do not need to be modified. You need to modify the attributes of the ringing current

Item		Remarks
		according to the local standards only when the default ringing current attributes do not meet the local standards.

Procedure

1.21.1 Configuring an MG Interface

This topic describes how to configure an MG interface for implementing the communication between the MA5600T/MA5603T/MA5608T (AG) and the MGC.

Context

- The MA5600T/MA5603T/MA5608T communicates with the MGC through either the H.248 or the MGCP protocol. One MA5600T/MA5603T/MA5608T can run only one protocol.
- One MA5600T/MA5603T/MA5608T supports up to eight MG interfaces. If a CKMC daughter board is configured in the MA5600T/MA5603T/MA5608T, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently. Each MG interface can be configured with the attributes (such as authentication and ringing mapping) independently.

Procedure

1.21.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.21.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.21.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.

2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.

2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
```

```
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

1.21.1.3 Adding an MG Interface

This topic describes how to add an MG interface, through which the Access node can communicate with the MGC.

Context

- One Access node supports a maximum of 8 MG interfaces. If a CKMC daughter board is configured in the Access node, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently.
- The configuration of the attributes of an MG interface is valid only to the MG interface.

Procedure

- Add an MG interface that supports H.248.
  a. Run the **display protocol support** command to query the current system protocol.
     - If the current system protocol is H.248, go to h.
     - If the current system protocol is MGCP, go to b.
  b. Run the **display if-mgcp all** command to query whether an MG interface that supports MGCP exists.
     - If such an MG interface does not exist, go to e.
     - If such an MG interface exists, go to c.
  c. Delete all configuration data of this MG interface, and then run the **shutdown(mgcp)** command to disable the MG interface.

NOTICE

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

  d. Run the **undo interface mgcp** command to delete the MG interface.
  e. Run the **protocol support** command to change the system protocol to H.248.
  f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.
  g. After the system is restarted, log in to the system, and enter the global config mode.
  h. Run the **interface h248** command to add an MG interface that supports H.248.
  i. Run the **if-h248 attribute** command to configure the attributes of the MG interface according to the data plan.

   j. Run the **display if-h248 attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

- Add an MG interface that supports MGCP.

   a. Run the **display protocol support** command to query the current system protocol.

      ■ If the current system protocol is MGCP, go to h.

      ■ If the current system protocol is H.248, go to b.

   b. Run the **display if-h248 all** command to query whether an MG interface that supports H.248 exists.

      ■ If such an MG interface does not exist, go to e.

      ■ If such an MG interface exists, go to c.

   c. Delete all configuration data of this MG interface, and then run the **shutdown(h248)** command to disable the MG interface.

⚠ CAUTION

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

   d. Run the **undo interface h248** command to delete the MG interface.

   e. Run the **protocol support** command to change the system protocol to MGCP.

   f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.

   g. After the system is restarted, log in to the system, and enter the global config mode.

   h. Run the **interface mgcp** command to add an MG interface that supports MGCP.

   i. Run the **if-mgcp attribute** command to configure the attributes of the MG interface according to the data plan.

   j. Run the **display if-mgcp attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

----End

Example

Assume that the MG interface ID is 0 and the H.248 protocol is used for interconnecting with the MGC. To add the MG interface, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#interface h248 0
  Are you sure to add MG interface?(y/n)[n]:y
```

Assume that the MG interface ID is 0 and the MGCP protocol is used for interconnecting with the MGC. The current system protocol, however, is the H.248 protocol. It is confirmed that the H.248 interface exists in the system but is not in use. To add MG interface 0, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#display if-h248 all
```

```
  ----------------------------------------------------------------------------
  MGID    Trans     State     MGPort MGIP          MGCPort MGCIP/DomainName
  ----------------------------------------------------------------------------
  0       -        Closed     -     -              -      -
  ----------------------------------------------------------------------------
huawei(config)#undo interface h248 0
  Are you sure to del MG interface?(y/n)[n]:y
huawei(config)#protocol support mgcp
huawei(config)#save
huawei(config)#reboot system
  Please check whether data has saved, the unsaved data will lose if reboot
system, are you sure to reboot system? (y/n)[n]:y
```

After the system is restarted, re-log in to the system.

```
huawei(config)#display protocol support
System support MGCP protocol
huawei(config)#interface mgcp 0
  Are you sure to add MG interface?(y/n)[n]:y
```

1.21.1.4 (Optional) Configuring the Digitmap of an MG Interface

This topic describes how to configure the digitmaps for certain special applications such as emergency calls and emergency standalone. The digitmaps for common phone services are issued to the AG by the MGC, and therefore such digitmaps do not need to be configured on the AG. If the services such as emergency calls and emergency standalone are not required, the digitmaps do not need to be configured.

Prerequisites

NOTICE

The digitmap configuration is relatively complicated. The information such as the meanings and usage of the characters in a digitmap is defined in the protocol, and is not described here. This topic provides only some basic information. It is recommended that you refer to the digitmap description in ITU-T H248.1 (applicable to H.248) or RFC 3435 (applicable to MGCP) before configuring the digitmap.

Context

- A digitmap is a set of digit collection descriptors. It is a dialing scheme resident in the MG and is used for detecting and reporting digit events received on a termination. The digitmap is used to improve the efficiency of the MG in sending the callee number. That is, if the callee number matches a dialing scheme defined by the digitmap, the MG sends the callee number collectively in a message.
- A digitmap consists of strings of digits with certain meanings. When the received dialing sequence matches one of the strings, the digits are collected completely.
- To configure the emergency standalone function, you must configure the internal digitmap.

The H.248-based MG interface supports the following types of digitmaps:

- Internal digitmap

- Emergency digitmap
- Emergency call digitmap (due to call restriction in case of an overload)
- Automatic redial digitmap of the card service

Table 1-36 provides the valid characters in the strings and their meanings in the H.248 protocol. For details about the digitmap in the H.248 protocol, refer to ITU-T H248.1, which provides a better guide to the digitmap configuration.

Table 1-36 Digitmap format in the H.248 protocol

Digit or Character	Description
0-9	Indicate dialed digits 0-9.
A-D	-
E	Indicates * in the DTMF mode.
F	Indicates for # in the DTMF mode.
X	Indicates for a wildcard, indicating any digit from 0 to 9.
S	Indicates the short timer. When detecting the timer timeout, the system reports a phone number digit by digit if the phone number is detected after the dialing scheme is matched.
L	Indicates the long timer. When detecting the timer timeout, the system reports a phone number digit by digit if the phone number is detected after the dialing scheme is matched.
Z	Indicates the duration modifier, which indicates a dialing event of a long duration. It is before the event character with a fixed location. When the event duration exceeds the threshold, the dialing event fills the location.
.	Indicates that there can be multiple digits (including 0) or characters before it.
|	Used to separate the strings and indicates that each string is an optional dialing scheme.
[]	Indicates that one digit or string can be selected from the options.

The MGCP-based MG interface supports the following types of digitmaps:

- Emergency call digitmap (due to call restriction in case of an overload)
- Automatic redial digitmap of the card service

Table 1-37 provides the valid characters in the strings and their meanings in the MGCP protocol.

Table 1-37 Digitmap format in the MGCP protocol

Digit or Character	Description

Digit or Character	Description
0-9	Indicate dialed digits 0-9.
A-D	-
X	Indicates for a wildcard, indicating any digit from 0 to 9.
T	Indicates that when detecting the timer timeout, the system reports a phone number digit by digit if the phone number is detected after the dialing scheme is matched.
*	Indicates * in the DTMF mode.
#	Indicates for # in the DTMF mode.
.	Indicates that there can be multiple digits (including 0) or characters before it.
|	Used to separate the strings and indicates that each string is an optional dialing scheme.
[]	Indicates that one digit or string can be selected from the options.

Procedure

- When the system protocol is H.248, perform the following operations to configure the digitmap.
  a. In the global config mode, run the **interface h248** command to enter the H.248 mode.
  b. Run the **digitmap set** command to configure the digitmap listed in the data plan.
  c. (Optional) Run the **digitmap-timer** command to configure the digitmap timer.

     Generally, the digitmap timer is issued by the MGC. In this case, the issued digitmap timer prevails regardless of whether a timer is configured on the AG. When the MGC does not issue the digitmap timer and the default digitmap timer does not meet the service requirements, you can configure the digitmap timer in this step.
  d. Check whether the configuration of the digitmap timer is the same as that in the data plan.
     - Run the **display digitmap** command to check whether the digitmap is configured correctly.
     - Run the **display digitmap-timer** command to check whether the digitmap timer is configured correctly.
- When the system protocol is MGCP, perform the following operations to configure the digitmap.
  a. In the global config mode, run the **interface mgcp** command to enter the MGCP mode.
  b. Run the **digitmap set** command to configure the digitmap listed in the data plan.
  c. (Optional) Run the **digitmap-timer** command to configure the digitmap timer.

Generally, the digitmap timer is issued by the MGC. In this case, the issued digitmap timer prevails regardless of whether a timer is configured on the AG. When the MGC does not issue the digitmap timer and the default digitmap timer does not meet the service requirements, you can configure the digitmap timer in this step.

d. Check whether the configuration of the digitmap timer is the same as that in the data plan.

   - Run the **display digitmap** command to check whether the digitmap is configured correctly.
   - Run the **display digitmap-timer** command to check whether the digitmap timer is configured correctly.

----End

Example

Assume that the inner digitmap of the H.248-based MG interface is configured. According to the data plan, the inner digitmap format is 1234xxxx. The digitmap timer is not configured because it is issued by the MGC. To configure the inner digitmap, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#digitmap set inner 1234xxxx
huawei(config-if-h248-0)#display digitmap
  ----------------------------------------------------------------------
  Inner digitmap                                  : 1234xxxx
  Emergency digitmap                               : -
  Urgent digitmap (for overload or bandwidth restrict)    : -
  Dualdial digitmap for card service                  : -
  ----------------------------------------------------------------------
```

1.21.1.5 (Optional) Configuring the Software Parameters of an MG Interface

The software parameters of an MG interface mainly define certain common service attributes of the MG interface. After the configuration of the software parameters of an MG interface, the software parameters take effect immediately and the configuration is valid only to the MG interface.

Context

There are 34 software parameters (numbered from 0 to 33) of an MG interface that supports H.248. Table 1-38 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-38 Software parameters of an MG interface that supports H.248

Parameter	Description	Default Setting
2	Indicates whether the MG interface supports dual homing. To configure an MG interface to or not to support dual	0: indicates that dual homing is not supported.

Parameter	Description	Default Setting
	homing, use this parameter. If the MG interface does not support dual homing (value: 0), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC. If the MG interface supports dual homing and auto-switching (value: 2), even if the MG interface has registered with the secondary MGC, the MG interface can automatically switch back to the primary MGC when the primary MGC recovers.	
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	0: indicates that a wildcard is used.
6	Indicates whether the MG interface supports device authentication. To configure an MG interface to or not to support authentication, use this parameter. After the device authentication is supported, run the **auth(h248)** command to configure the authentication parameters, and then run the **reset(h248)** command to reset	1: indicates that device authentication is not supported.

Parameter	Description	Default Setting
	the MG interface. In this way, the MGC can manage the security of the MGs and avoid illegal registration with the MGC.	
7	Indicates whether the MG interface supports security header. To configure an MG interface to or not to support security header, use this parameter.	1: indicates that security header is not supported.
11	Indicates whether the MG interface supports emergency standalone. To configure whether an MG interface supports emergency standalone, use this parameter. If the MG interface supports emergency standalone, the users on the MG interface can make phone calls even if the MG fails to communicate with the MGC.	0: indicates that no call is permitted.
13	Digitmap matching mode	2: indicates the minimum matching.
15	Indicates whether the function of filtering media streams by source port is enabled on an MG interface. To enable or disable the function of filtering media streams by source port on an MG interface, use this parameter. When the function of filtering media streams by source port is enabled on the MG interface, only the media streams from the specific ports can be received.	0: indicates that media streams are not filtered by source port.
16	Indicates the length of the timer for filtering the media stream source port of the MG interface. To configure the length of the timer for filtering the media stream source port of an MG	5s

Parameter	Description	Default Setting
	interface, use this parameter. When an MG interface does not filter the source port, the MG interface automatically filters the source port if the filtering timer times out.	
22	Indicates the type of the prompt tone that is played after the communication between the MG and the MGC is interrupted. To configure the type of the prompt tone after the communication between the MG and the MGC is interrupted, use this parameter.	0: indicates the busy tone.
23	Indicates the length of the timer for synchronizing the port status. To configure the length of the timer for synchronizing the port status, use this parameter.	35s
24	Indicates the maximum value of the Real-Time Transport Protocol (RTP) termination ID.	-
25	Indicates the maximum random value for the protection against avalanche of the H.248 interface.	-
26	Indicates the type of local blocking play tone.	0: indicates the busy tone.
27	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
28	Indicates the duration of the howler tone.	60s
29	Indicates the duration of message waiting tone.	3s
30	Indicates the time limit of the alarm for extra long call.	60 minutes
31	Indicates whether to report the alarm for extra long call.	1: indicates that the alarm is not reported.
32	Min. auto registration interval of remotely-blocked port(s).	1800s

Parameter	Description	Default Setting
33	Whether MG heartbeat is shut down.	1: No, heartbeat is enabled

There are 17 software parameters (numbered from 0 to 16) of an MG interface that supports MGCP. Table 1-39 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-39 Software parameters of an MG interface that supports MGCP

Parameter	Description	Default Setting
1	Indicates whether the ongoing call is maintained if the communication between the MG interface and the MGC is abnormal. To maintain the ongoing call when the communication between the MG interface and the MGC is abnormal, use this parameter.	1: disconnects all the calls at once.
2	Indicates whether the MG interface supports dual homing. To configure whether an MG interface supports dual homing, use this parameter. If the MG interface does not support dual homing (value: 1), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC.	0: indicates that dual homing is supported.
3	Indicates whether the heartbeat message between the MG and the MGC is disabled. To configure whether the heartbeat message between the MG and the MGC is disabled, use this parameter.	1: indicates that the heartbeat message is not disabled.
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the	0: indicates that a wildcard is used.

Parameter	Description	Default Setting
	registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	
5	Indicates the MGC type. To select the MGC of a different type, use this parameter.	0
6	Indicates the maximum time threshold for responding to the heartbeat messages. To configure the maximum times for transmitting the heartbeat message continuously, use this parameter.	3
7	Indicates whether to report the heartbeat with the MG as an endpoint. To configure whether to report the heartbeat with the MG as an endpoint, use this parameter.	0: indicates that reporting the heartbeat with the MG as an endpoint is not supported.
10	Indicates the point-to-point (P2P) fault reporting. To configure whether the MG interface supports the P2P fault reporting from the ISDN terminals, use this parameter.	0: indicates that the P2P fault is reported.
11	Indicates the point-to-multipoint (P2MP) fault reporting. To configure whether the MG interface supports the P2MP fault reporting from the ISDN terminals, use this parameter.	1: indicates that the P2MP fault is not reported.

Parameter	Description	Default Setting
12	Indicates the type of local blocking play tone.	0: indicates the busy tone.
13	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
14	Indicates the RTP filtering switch of the MGCP interface. To configure whether the RTP filtering function is enabled, use this parameter. When the RTP filtering function is enabled, only the media stream from the specific ports can be received.	1: indicates that the RTP filtering function is not enabled.
15	Indicates the duration of the howler tone.	60s
16	Whether the timer symbol "T" follows the number string reported by the signaling.	0: Yes

Procedure

- When the system protocol is H.248, perform the following operations to configure the software parameters of an MG interface.
  a. In the global config mode, run the **interface h248** command to enter the MG interface mode.
  b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.
  c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.
- When the system protocol is MGCP, perform the following operations to configure the software parameters of an MG interface.
  a. In the global config mode, run the **interface mgcp** command to enter the MG interface mode.
  b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.
  c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 11 of H.248-based MG interface 0 to 1 so that the MG interface supports emergency standalone and allows only internal calls, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-software parameter 11 1
huawei(config-if-h248-0)#display mg-software parameter 11
  ------------------------------------------------
  Interface Id:0         para index:11  value:1
  ------------------------------------------------
 APPENDIX:
  ------------------------------------------------
   Interface software parameter name:
   11: Stand alone flag
       0: None
       1: Inner
       2: Emergency
       3: Both
```

1.21.1.6 (Optional) Configuring the Ringing Mode of an MG Interface

This topic describes how to configure the ringing mode of an MG interface to meet different user requirements.

Procedure

- If the system protocol is H.248, perform the following operations to configure the ringing mode of an MG interface.
  a. Check whether the preset ringing mode in the system meets the requirements according to the Usage Guidelines of the **mg-ringmode add** command.
     - If the preset ringing mode meets the requirements, go to c.
     - If the preset ringing mode does not meet the requirements, proceed to b.

  b. In the global config mode, run the **user defined-ring modify** command to configure the break-make ratio of user-defined ringing mode to form a ringing mode that meets the user requirements.

NOTICE

- After configuring the user-defined ringing mode, you need to reset the corresponding service board to make the configuration data take effect. Therefore, the user of the service board can use the new user-defined ringing mode. If the service board has been in service for a period, evaluate the impact on the online users before resetting the service board, and then determine whether to perform this operation.
- The system supports 16 user-defined ringing modes, which can be modified but cannot be added or deleted.

  c. Run the **interface h248** command to enter the H.248 mode.

  d. Run the **mg-ringmode add** command to add a ringing mapping.

NOTICE

1. The parameter *mgcpara* on the MG must be the same as the parameter *mgcpara* on the MGC.
2. User-defined ringing modes 0 to 15 correspond to cadence ringing modes 128 to 143 respectively, and correspond to initial ringing modes 144 to 159 respectively. For example, if the cadence ringing mode is 128, user-defined ringing mode 0 is selected. If the initial ringing mode is 144, user-defined ringing mode 0 is selected.

e. Run the **display mg-ringmode attribute** command to check whether the ringing mapping is the same as that in the data plan.

- If the system protocol is MGCP, perform the following operations to configure the ringing mode of an MG interface.

  a. According to the Usage Guidelines of the **mg-ringmode add** command, check whether the preset ringing mode in the system meets the requirements.

     - If the preset ringing mode meets the requirements, go to c.
     - If the preset ringing mode does not meet the requirements, proceed to b.

  b. In the global config mode, run the **user defined-ring modify** command to configure the user-defined ringing mode.

NOTICE

After configuring the user-defined ringing mode, you need to reset the corresponding service board to make the configuration data take effect. Therefore, the user of the service board can use the new user-defined ringing mode. If the service board has been in service for a period, evaluate the impact on the online users before resetting the service board, and then determine whether to perform this operation.

  c. Run the **interface mgcp** command to enter the MGCP mode.

  d. Run the **mg-ringmode add** command to add a ringing mapping.

NOTICE

The parameter *mgcpara* on the MG must be the same as the parameter *mgcpara* configured on the MGC.

  e. Run the **display mg-ringmode attribute** command to check whether the ringing mapping is the same as that in the data plan.

----End

Example

Assume that the MG interface ID is 0, the peer parameter ID issued by the MGC is 0, the cadence ringing is 1:4 (the value of the corresponding parameter *cadence* is 0), and the initial ringing is 1:2 (the value of the corresponding parameter *initialring* is 17). To configure the ringing mode of MG interface 0, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-ringmode add 0 0 17
huawei(config-if-h248-0)#display mg-ringmode attribute
{ <cr>|mgcpara<U><0,15> }:

  Command:
        display mg-ringmode attribute
  -------------------------------------------------------
   MGID      PeerPara   CadenceRing   InitialRing
  -------------------------------------------------------
   0        0         0           17
  -------------------------------------------------------
```

Assume that the MG interface ID is 0, the peer parameter ID issued by the MGC is 0, the break-make ratio of user-defined ringing mode 0 is 0.4sec On, 0.2sec Off, 0.4sec On, 2.0sec Off, and the initial ringing and the cadence ringing use user-defined ringing mode 0 (the values of the corresponding parameters *cadence* and *initialring* are 128 and 144 respectively). To configure the ringing mode of MG interface 0, do as follows:

```
huawei(config)#user defined-ring modify 0 para1 400 para2 200 para3 400 para4 2000
  Note: Please reset the service board to make configured parameter be valid
huawei(config)#display user defined-ring
  --------------------------------------------------
  RingType Para1 Para2 Para3 Para4 Para5 Para6
  --------------------------------------------------
  0       400   200   400   2000    0     0
  1         0    0     0     0     0     0
  2         0    0     0     0     0     0
  3         0    0     0     0     0     0
......
  14        0    0     0     0     0     0
  15        0    0     0     0     0     0
  --------------------------------------------------
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-ringmode add 1 128 144
huawei(config-if-h248-0)#display mg-ringmode attribute
{ <cr>|mgcpara<U><0,15> }:
{ <cr>|mgcpara<U><0,15> }:

  Command:
        display mg-ringmode attribute
  -------------------------------------------------------
   MGID      PeerPara   CadenceRing   InitialRing
  -------------------------------------------------------
   0        1        128         144
  -------------------------------------------------------
```

1.21.1.7 (Optional) Configuring the TID Format of an MG Interface

The TID format determines how various user terminal IDs on the MG interface are generated.

Prerequisites

NOTICE

The configuration of the TID format is relatively complicated. The information such as the syntax rules with which the terminal prefix must comply and the requirements for the character string of the TID template is defined in the protocol, and is not described here. This topic describes only some basic information. It is recommended that you refer to the TID description in ITU-T H248.1 (applicable to H.248) or RFC 3435 (applicable to MGCP) before configuring the TID format.

Context

The TID format consists of the terminal prefix and the TID template. The TID template defines the generation mode of the TID excluding the terminal prefix. A TID consists of a terminal prefix and a character string generated by a TID template.

The TID templates that are bound to various types of users on the MG interface determine whether the users support terminal layering.

- If the parameter list of the TID template includes only keyword "G", the TID template is used by the non-layering users. Users bound with this template do not support terminal layering.
- If the parameter list of the TID template includes only keywords "F", "S", "P", "B" ("B" is not available to PSTN users), the TID template is used by the layering users. Users bound with this template support terminal layering.

NOTE

The meaning of each keyword is as follows:

- F indicates the subrack ID.
- S indicates the slot ID.
- P indicates the port ID.
- B indicates the B channel ID (only for ISDN terminals).
- G indicates the general permanent termination index.
- R indicates the RTP ephemeral termination index (only for the RTP ephemeral termination, which exists only when the system protocol is H.248. This termination is not involved unless special remarks are provided.)

When adding a user that supports terminal layering, you cannot and do not have to specify the parameter **terminalid** because the system automatically allocates a terminal ID. When adding a user that does not support terminal layering, you must specify the parameter **terminalid**.

You can run the **display tid-format(h248)** command or **display tid-format(mgcp)** command to query the TID formats of various types of users on the current MG interface. In the query result, **template-index** indicates the index of the TID template that is bound to the type of users. Then, run the **display tid-template** command to check whether the TID template supports the layering configuration. Hence, you can check whether the user supports terminal layering.

Precaution

- There are 19 default TID templates (templates 0-18) in the system. The default TID templates can be referenced, but cannot be added, modified, or deleted.
- The configuration of terminal layering on the MG must be the same as that on the MGC.
- If certain type of terminals exists on an interface and the interface is not disabled, the terminal prefix of this type of terminals cannot be modified.

- If a certain type of terminals exists on an interface, the TID format (including the terminal prefix and the index of the TID template) of this type of terminals cannot be changed.
- The terminal prefix must comply with the following syntax rules: The prefix must not exceed 64 characters. Only letters, digits, "_", and "/" are the characters allowed for input. The first character must be a letter, and the last character must not be a digit.
- The length of the TID, which is generated by combining the TID template and the terminal prefix that you configured, must not exceed 64 characters.

Procedure

- If the system protocol is H.248, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface h248** command to enter the H.248 mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).

NOTICE

The terminal prefix of PSTN, BRA and PRA must be all identical or unique.

  - In the H.248 mode, run the **tid-format rtp** command to configure the TID template and the terminal prefix of the RTP ephemeral termination.
  - In the H.248 mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
  - In the H.248 mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
  - In the H.248 mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.

  e. Run the **display tid-format(h248)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.
- If the system protocol is MGCP, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface mgcp** command to enter the MGCP mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).

  - In the MGCP mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
  - In the MGCP mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
  - In the MGCP mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.

e. Run the **display tid-format(mgcp)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.

----End

Example

Assume that in the H.248 mode, the terminal prefix of the PSTN user on MG interface 1 is aln/, and the layering TID template 3 is used. To add a PSTN user on port 0/2/0 and check whether the system automatically allocates a TID generated according to the template, do as follows:

```
huawei(config)#display tid-template 3//Query the information about TID template 3
  -------------------------------------------------
  Index     : 3
  Format    : %u/%u/%u
  Para-list  : F+1,S+1,P+1  //The parameter list of the TID template includes keyword
"F", "S",
               //and "P", which indicates that this template supports terminal layering.
  Name      : Aln_Not_Fixed_1
  -------------------------------------------------
huawei(config)#interface h248 1
huawei(config-if-h248-1)#tid-format pstn prefix aln template 3
huawei(config-if-h248-1)#quit
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser add 0/2/0 1
huawei(config-esl-user)#display mgpstnuser 0/2/0
{ <cr>|endframeid/slotid/portid<S><1,15> }:

  Command:
        display mgpstnuser 0/15/0
  ----------------------------------------------------------------------
  F  /S /P   MGID    TelNo          Priority PotsLineType TerminalID
  ----------------------------------------------------------------------
  0  /2 /0   1       -             Cat3    DEL        aln/1/3/1
           //The system allocates the terminal ID according to the TID format.
  ----------------------------------------------------------------------
```

1.21.1.8 Enabling an MG Interface

Enabling an MG interface is to reset an MG interface to make the MG interface register with the MGC (or to make the modified attributes of the MG interface take effect) after the configuration of the MG interface is complete, so that the MG interface can work in the normal state.

Precaution

CAUTION

For the MG interface that has been in service, this operation interrupts the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation.

Procedure

- Enable the MG interface that adopts the H.248 protocol.
  a. Run the **interface h248** command to enter the H.248 mode.
  b. Run the **reset coldstart** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-h248 all** command to check whether the MG interface is in the normal state.
- Enable the MG interface that adopts the MGCP protocol.
  a. Run the **interface mgcp** command to enter the MGCP mode.
  b. Run the **reset** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-mgcp all** command to check whether the MG interface is in the normal state.

----End

Example

To enable H.248-based MG interface 0, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-h248-0)#quit
huawei(config)#display if-h248 all
-------------------------------------------------------------------------------
MGID    Trans    State        MGPort MGIP         MGCPort MGCIP/DomainName
-------------------------------------------------------------------------------
0      UDP      Normal        2944   10.10.10.11   2944   10.10.20.11
-------------------------------------------------------------------------------
```

To enable MGCP-based MG interface 0, do as follows:

```
huawei(config)#interface mgcp 0
huawei(config-if-mgcp-0)#reset
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-mgcp-0)#quit
huawei(config)#display if-mgcp all
  -------------------------------------------------------------------------
  MGID    State        MGPort MGIP          MGCPort MGCIP/DomainName
  -------------------------------------------------------------------------
  0      Normal        2727   10.10.10.11   2727    10.10.20.11
```

```
1      Wait ack      2527   10.10.10.12    2727   10.10.20.12
--------------------------------------------------------------------------
```

1.21.2 Configuring the VoIP PSTN User

After an MG interface is configured, you can add plain old telephone service (POTS) users on the MG interface to implement the VoIP PSTN service.

Procedure

1.21.2.1 Configuring the PSTN User Data

This topic describes how to configure the PSTN user data (the same as the corresponding data on the MGC) on the MG interface to provide the POTS terminal with the access to the network for VoIP service.

Prerequisites

The POTS service board must be inserted into the planned slot, and the RUN ALM LED of the board must be green and must be on for 1s and off for 1s repeatedly.

NOTE

You can add a service board in two ways (see the Usage Guideline of the **board add** command). It is recommended that you insert the service board into the planned slot and then confirm the board.

Procedure

Step 1 In the global config mode, run the **board confirm** command to confirm the service board.

Step 2 Add a PSTN user.

1. In the global config mode, run the **esl user** command to enter the ESL user mode.
2. Run the **mgpstnuser add** or **mgpstnuser batadd** command to add a PSTN user or add PSTN users in batches.

NOTICE

- When you add a PSTN user, the terminal ID must be configured and must be different from the terminal ID of an existing PTSN user if the TID template with which the PSTN user on the MG interface is bound is not a layering template.
- When you add a PSTN user, the configuration of the terminal ID is not required and the system automatically allocates the terminal ID if the TID template with which the PSTN user on the MG interface is bound is a layering template.
- When adding a PSTN user, you can configure the phone number (parameter *telno*). The phone number configured, however, can be used only as the paging number for emergency standalone. Phone numbers for normal call services are allocated by the MGC. It is recommended that the phone number configured here be the same as the phone number allocated by the MGC. In addition, the phone number must be unique in the MG. This is to avoid the number conflict that may occur when emergency standalone is enabled. If this parameter is not set, the phone number is null by default.
- For details about the relation between the TID template and the terminal layering, see the Background Information in 1.21.1.7 (Optional) Configuring the TID Format of an MG Interface.

3. Run the **display mgpstnuser** command to check whether the PSTN user data is the same as that in the data plan.

Step 3 (Optional) Configure the attributes of the PSTN user.

The attributes of a PSTN user need to be configured when the default settings are not consistent with the actual application.

1. Run the **mgpstnuser attribute set** or **mgpstnuser attribute batset** command to configure the attributes of the PSTN user.
2. Run the **display mgpstnuser attribute** command to check whether the attributes of the PSTN user are the same as those in the data plan.

----End

Example

Assume that the phone numbers of 32 PSTN users are 83110000-83110031, the **terminalid** values are 0-31 (the TID template to which the PSTN users under the MG interface are bound does not support layering and **terminalid** should be allocated manually), and the default values are used for other attributes. To add the 32 PSTN users in slot 0/2 under MG 0 in batches, do as follows:

```
huawei(config)#board confirm 0/2
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser batadd 0/2/0 0/2/31 0 terminalid 0 telno 83110000
huawei(config-esl-user)#display mgpstnuser 0 0 32
  ------------------------------------------------------------------------
  F  /S /P    MGID    TelNo          Priority PotsLineType TerminalID
  ------------------------------------------------------------------------
  0  /2 /0    0       83110000       Cat3     DEL          A0
  0  /2 /1    0       83110001       Cat3     DEL          A1
  ......
  0  /2 /30   0       83110030       Cat3     DEL          A30
  0  /2 /31   0       83110031       Cat3     DEL          A31
  ------------------------------------------------------------------------
```

1.21.2.2 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  ------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
  ------------------------------------------------------------------------
```

1.21.2.3 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1
```

```
Command:
      display oversea parameters 1
  ------------------------------------------------------------------------
Parameter name index: 1    Parameter value: 100
Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
-----------------------------------------------------------------------
```

1.21.2.4 (Optional) Configuring the Attributes of a PSTN Port

This topic describes how to configure the attributes of a PSTN port to ensure that the PSTN port can meet the actual application requirements.

Context

The MA5600T/MA5603T/MA5608T supports the following attributes of a PSTN port:

- Physical attributes (including whether to support the polarity reversal pulse, whether to support the port locking, and dialing mode). For details about how to configure the physical attributes of a PSTN port, see **pstnport attribute set**.
- Electrical attributes (including the impedance and the current). For details about how to configure the electrical attributes of a PSTN port, see **pstnport electric set**.
- KC attributes (including the KC accounting mode and the valid voltage). For details about how to configure the KC attributes of a PSTN port, see **pstnport kc set**.

Procedure

Step 1 In the global config mode, run the **pstnport** command to enter the PSTN port mode.

Step 2 Run the **pstnport attribute batset** or **pstnport attribute set** command to configure the physical attributes of the PSTN port.

Step 3 Run the **pstnport electric batset** or **pstnport electric set** command to configure the electrical attributes of the PSTN port.

Step 4 Run the **pstnport kc batset** or **pstnport kc set** command to configure the KC attributes of the PSTN port.

Step 5 Check whether the attribute configuration of the PSTN port is the same as that in the data plan.

- Run the **display pstnport attribute** command to query the physical attributes of the PSTN port.
- Run the **display pstnport electric** command to query the electrical attributes of the PSTN port.
- Run the **display pstnport kc** command to query the KC attributes of the PSTN port.

----End

Example

To configure the 32 PSTN ports of the board in slot 0/3 to support the polarity reversal accounting, do as follows:

```
huawei(config)#pstnport
huawei(config-pstnport)#pstnport attribute batset 0/3/0 0/3/31 reverse-pole-pulse
```

```
enable
huawei(config-pstnport)#display pstnport attribute 0/3
  --------------------------------------------------------------------
  F  /S  /P            0/3 /0
  ReversePolepulse       Enable
  PulseLevel            100(ms)
  PolarityReverseMode    Hard-polarity-reverse
  Dial-Mode             DTMF-Pulse-Both
  LineLock             Enable
  NlpMode              Nlp normal mode
  PolarityReverseWhenCLIP Disable
  PulsePeriodUpperLimit   200(ms)
  PulsePeriodLowerLimit   50(ms)
  PulseDurationUpperLimit 90(ms)
  PulseDurationLowerLimit 30(ms)
  PulsePauseUpperLimit   90(ms)
  PulsePauseLowerLimit   30(ms)
  OffhookTime(Idle)      80(ms)
  OffhookTime(Ring)      200(ms)
  OffhookTime(Fsk)       50(ms)
  --------------------------------------------------------------------
  F  /S  /P            0/3 /1
  ReversePolepulse       Enable
  PulseLevel            100(ms)
  PolarityReverseMode    Hard-polarity-reverse
  Dial-Mode             DTMF-Pulse-Both
  LineLock             Enable
  NlpMode              Nlp normal mode
  PolarityReverseWhenCLIP Disable
  PulsePeriodUpperLimit   200(ms)
  PulsePeriodLowerLimit   50(ms)
  PulseDurationUpperLimit 90(ms)
  PulseDurationLowerLimit 30(ms)
  PulsePauseUpperLimit   90(ms)
  PulsePauseLowerLimit   30(ms)
  OffhookTime(Idle)      80(ms)
  OffhookTime(Ring)      200(ms)
  OffhookTime(Fsk)       50(ms)
  ----------------------------------------------------------------------------
  F  /S  /P            0/3 /31
  ReversePolepulse       Enable
  PulseLevel            100(ms)
  PolarityReverseMode    Hard-polarity-reverse
  Dial-Mode             DTMF-Pulse-Both
  LineLock             Enable
  NlpMode              Nlp normal mode
  PolarityReverseWhenCLIP Disable
  PulsePeriodUpperLimit   200(ms)
  PulsePeriodLowerLimit   50(ms)
  PulseDurationUpperLimit 90(ms)
  PulseDurationLowerLimit 30(ms)
  PulsePauseUpperLimit   90(ms)
  PulsePauseLowerLimit   30(ms)
  OffhookTime(Idle)      80(ms)
  OffhookTime(Ring)      200(ms)
  OffhookTime(Fsk)       50(ms)
  ---------------------------------------------------------------------------
```

NOTE

When a call begins and ends, the MG shows the start time and the end time based on the polarity reversal on the subscriber line. The billing terminal that supports the polarity reversal accounting function, such as a charging phone set, implements the polarity reversal accounting function based on the start time and the end time of a call.

1.21.2.5 (Optional) Configuring the Attributes of the Ringing Current

You can adjust the ringing volume and ringing tone by modifying the attributes of the ringing current. In general, the attributes of the ringing current do not need to be modified. You need to modify the attributes of the ringing current according to the local standards only when the default ringing current attributes do not meet the local standards.

Context

The attributes of the ringing current include the following two parameters:

- Ringing current frequency: A higher frequency indicates a sharper ringing tone. The default ringing current frequency is 25 Hz.
- AC amplitude (AC voltage): A greater amplitude indicates a louder ringing tone. The default AC amplitude is 75 Vrms.

Procedure

Step 1 In the global config mode, run the **voip** command to enter the VoIP mode.

Step 2 Run the **ring attribute set** command to configure the attributes of the ringing current according to the data plan.

Step 3 Run the **display ring attribute** command to check whether the attributes of the ringing current are the same as those in the data plan.

----End

Example

To set the ringing current frequency to 50 Hz (parameter value 2), AC amplitude to 50 Vrms (parameter value 2), do as follows:

```
huawei(config-voip)#ring attribute set frequency 2 acamplitude 2
huawei(config-voip)#display ring attribute
  ringing current frequency  : 50HZ
  ringing current acamplitute: 50VRMS
```

1.22 Configuring the VoIP ISDN BRA Service (H.248-based)

This topic describes how to configure the VoIP ISDN BRA service on an IP network. When the Access node uses the H.248 protocol, the device supports the access of the ISDN BRA user. ISDN technology provides end-to-end (E2E) digital connection and supports multiple types of voice and non-voice telecom services.

Prerequisites

According to the actual network, a route from the Access node to the MGC must be configured to ensure that the Access node communicates with the MGC normally.

Context

- The ISDN is integrated services digital network. Defined by the International Telegraph and Telephone Consultative Committee (CCITT), the ISDN technology provides two types of user-network interfaces based on the user-network interface reference model.
  - ISDN basic rate interface (BRI), which supports a rate of 144 kbit/s and provides two B channels (for carrying services) and one D channel (for transmitting call control signaling and maintenance and management signaling). The rate of B channels is 64 kbit/s and the rate of D channel is 16 kbit/s. The service carried on the ISDN BRI is the ISDN basic rate access (BRA) service.
  - ISDN primary rate interface (PRI), which supports a rate of 2.048 Mbit/s and provides 30 B channels and one D channel. The rates of the B channel and D channel are both 64 kbit/s. The service carried on the ISDN PRI is the ISDN primary rate access (PRA) service.

Precaution

The media gateway control protocol (MGCP) is a master/slave protocol. under which the MGC controls the AG to implement call connection and disconnection. The data on the AG for the interconnection with the MGC, such as the attributes of the MG interface and the attributes of the VoIP user, must be the same as the corresponding data on the MGC. Therefore, before configuring the VoIP service, you must contact MGC engineers to check and ensure that the interconnection data plan for the AG is consistent with the corresponding plan for the MGC.

Data preparation

Table 1-40 provides the data plan for configuring the H.248-based VoIP ISDN BRA service.

Table 1-40 Data plan for configuring the H.248-based VoIP ISDN BRA service

Item			Remarks
MG interface data (The data must be consistent with the data on the MGC.)	Parameters of the media stream and signaling stream	Upstream VLAN for media and signaling streams	It is used for the upstream VLAN of the VoIP service to be configured. Standard VLAN is recommended.
		Uplink port for media and signaling streams	It is used as the uplink port for the VoIP service to be configured.
		Media IP address and signaling IP address	These IP addresses must be contained in the media and signaling IP address pools. The media and signaling IP address pools consist of all the IP addresses of the Layer 3

Item			Remarks
			interface of the upstream VLAN for media and signaling streams.
		Default IP address of the MG	It is the next hop IP address from the Access node to the MGC.
	Attribute parameters of the MG interface **NOTE** There are many MG interface parameters. Only mandatory parameters are listed here. If the mandatory parameters are not configured, the MG interface cannot be started.	MG interface ID	It is the ID of the MG interface used by the VoIP service to be configured.
		Signaling port ID of the MG interface	It is the transport layer protocol port ID used for the signaling exchange between the Access node (AG) and the MGC. The default signaling port ID defined in H.248 is 2944 (text) and 2945 (binary).
		IP address of the MGC to which the MG interface belongs	The MGC can be specified by IP address or the domain name. The IP address is adopted here. When dual homing is not configured, you can configure the parameters of only the primary MGC. When dual homing is configured, you also need to configure the IP address and port ID of the secondary MGC.
		Port ID of the MGC to which the MG interface belongs	
		Domain name of the MGC to which the MG interface belongs	
		Coding mode of the MG interface	Only the **text** mode is supported.
		Transmission mode of the MG interface	The transmission mode is selected according to the requirements of the MGC. Generally, UDP is used.
		Domain name of the MG interface	It corresponds to the **domainName** parameter of the MG interface. When the H.248 protocol is used, this parameter must be configured if the **MIDType** parameter of the H.248 message is configured to **domainName**. Otherwise, the MG interface

Item			Remarks
			cannot be started. In other situations, this parameter is optional.
		Device name of the MG interface	It is supported by only the H.248 protocol, and it corresponds to the **deviceName** parameter of the MG interface that uses the H.248 protocol. This parameter must be configured if the **MIDType** parameter of the H.248 message is configured to **domainName**. Otherwise, the MG interface cannot be started. In other situations, this parameter is optional.
	Software parameters of the MG interface		Whether the MG interface supports the functions such as dual homing and emergency standalone is determined by the service requirements.
	Terminal ID (TID) format of the MG interface		The TID format determines the generation mode of various types of user terminals on an MG interface.
IUA link	IUA link set		The IUA link can be configured only after the IUA link set is configured.
	IUA link **NOTE** The local port ID, local IP address, remote port ID, and remote IP address of different links must not be completely same; otherwise, the service cannot be configured.	IUA link ID	It indicates the link for transmitting the signaling.
		IUA link set ID	-
		Local port ID	To activate the link normally, it must be the same as the remote port ID configured on the MGC.
		Local IP address	It must be the same as the remote IP address of the link configured on the MGC. In addition, the local IP addresses of the links that are in the same link set must be the same. (The IP address must exist in the media IP address pool.)
		Remote port ID	To activate the link normally, it must be the same as the local port ID configured on the MGC.

Item			Remarks
		Remote IP address	It must be the same as the local IP address of the link configured on the MGC. The SCTP protocol supports the multi-homing function. That is, one link can be configured with the IP addresses of multiple MGCs as the remote IP addresses. When one MGC is faulty, the link can be switched to other MGCs automatically. This ensures that the service is not affected. The Access node supports the configuration of the active remote IP address and standby remote IP address.
ISDN BRA user data (The data must be consistent with the data on the MGC.)	Slot that houses the ISDN BRA service board		-
	User data	Termination ID	If the TID format bound to the BRA user does not support terminal layering function, this parameter needs to be configured, and the configuration must be consistent with the configuration on the MGC.
		User priority	The user priority must be specified according to the service requirements. There are three categories of user priorities, which are as follows: • cat1: government1 (category 1 government user) • cat2: government2 (category 2 government user) • cat3: normal (common user, default) The priorities of cat1, cat2, and cat3 are in descending order. Without special requirements, the default cat3 is adopted.
		Interface ID	It indicates the interface for the BRA user data to pass through the MG and MGC. The configuration of this parameter must be consistent with the corresponding configuration on the MGC.

Item		Remarks
	System parameters	The system parameters including the oversea version flag and message waiting indication (MWI) mode need to be configured according to local standards to ensure that the response of the user terminal complies with local standards.
	Overseas parameters	The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to local standards to ensure that the response of the user terminal complies with local standards.
	Attributes of an ISDN BRA Port	The attributes such as the working mode, remote power supply status, and auto-deactivation status of the port can be configured. Modify such attributes only if there is a special requirement.

1.22.1 Configuring an MG Interface

This topic describes how to configure an MG interface for implementing the communication between the MA5600T/MA5603T/MA5608T (AG) and the MGC.

Context

- The MA5600T/MA5603T/MA5608T communicates with the MGC through either the H.248 or the MGCP protocol. One MA5600T/MA5603T/MA5608T can run only one protocol.
- One MA5600T/MA5603T/MA5608T supports up to eight MG interfaces. If a CKMC daughter board is configured in the MA5600T/MA5603T/MA5608T, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently. Each MG interface can be configured with the attributes (such as authentication and ringing mapping) independently.

Procedure

1.22.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.22.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.22.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

1.22.1.3 Adding an MG Interface

This topic describes how to add an MG interface, through which the Access node can communicate with the MGC.

Context

- One Access node supports a maximum of 8 MG interfaces. If a CKMC daughter board is configured in the Access node, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently.
- The configuration of the attributes of an MG interface is valid only to the MG interface.

Procedure

- Add an MG interface that supports H.248.
  a. Run the **display protocol support** command to query the current system protocol.
     - If the current system protocol is H.248, go to h.
     - If the current system protocol is MGCP, go to b.
  b. Run the **display if-mgcp all** command to query whether an MG interface that supports MGCP exists.
     - If such an MG interface does not exist, go to e.
     - If such an MG interface exists, go to c.

c. Delete all configuration data of this MG interface, and then run the **shutdown(mgcp)** command to disable the MG interface.

NOTICE

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

d. Run the **undo interface mgcp** command to delete the MG interface.

e. Run the **protocol support** command to change the system protocol to H.248.

f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.

g. After the system is restarted, log in to the system, and enter the global config mode.

h. Run the **interface h248** command to add an MG interface that supports H.248.

i. Run the **if-h248 attribute** command to configure the attributes of the MG interface according to the data plan.

j. Run the **display if-h248 attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

- Add an MG interface that supports MGCP.

a. Run the **display protocol support** command to query the current system protocol.

  - If the current system protocol is MGCP, go to h.
  - If the current system protocol is H.248, go to b.

b. Run the **display if-h248 all** command to query whether an MG interface that supports H.248 exists.

  - If such an MG interface does not exist, go to e.
  - If such an MG interface exists, go to c.

c. Delete all configuration data of this MG interface, and then run the **shutdown(h248)** command to disable the MG interface.

⚠CAUTION

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

d. Run the **undo interface h248** command to delete the MG interface.

e. Run the **protocol support** command to change the system protocol to MGCP.

f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.

g. After the system is restarted, log in to the system, and enter the global config mode.

h. Run the **interface mgcp** command to add an MG interface that supports MGCP.

i. Run the **if-mgcp attribute** command to configure the attributes of the MG interface according to the data plan.

j. Run the **display if-mgcp attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

----End

Example

Assume that the MG interface ID is 0 and the H.248 protocol is used for interconnecting with the MGC. To add the MG interface, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#interface h248 0
  Are you sure to add MG interface?(y/n)[n]:y
```

Assume that the MG interface ID is 0 and the MGCP protocol is used for interconnecting with the MGC. The current system protocol, however, is the H.248 protocol. It is confirmed that the H.248 interface exists in the system but is not in use. To add MG interface 0, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#display if-h248 all
  ----------------------------------------------------------------------------
  MGID    Trans    State    MGPort MGIP          MGCPort MGCIP/DomainName
  ----------------------------------------------------------------------------
  0       -       Closed    -     -            -      -
  ----------------------------------------------------------------------------
huawei(config)#undo interface h248 0
  Are you sure to del MG interface?(y/n)[n]:y
huawei(config)#protocol support mgcp
huawei(config)#save
huawei(config)#reboot system
  Please check whether data has saved, the unsaved data will lose if reboot
system, are you sure to reboot system? (y/n)[n]:y
```

After the system is restarted, re-log in to the system.

```
huawei(config)#display protocol support
System support MGCP protocol
huawei(config)#interface mgcp 0
  Are you sure to add MG interface?(y/n)[n]:y
```

1.22.1.4 (Optional) Configuring the Software Parameters of an MG Interface

The software parameters of an MG interface mainly define certain common service attributes of the MG interface. After the configuration of the software parameters of an MG interface, the software parameters take effect immediately and the configuration is valid only to the MG interface.

Context

There are 34 software parameters (numbered from 0 to 33) of an MG interface that supports H.248. Table 1-41 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-41 Software parameters of an MG interface that supports H.248

Parameter	Description	Default Setting
2	Indicates whether the MG interface supports dual homing. To configure an MG interface to or not to support dual homing, use this parameter. If the MG interface does not support dual homing (value: 0), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC. If the MG interface supports dual homing and auto-switching (value: 2), even if the MG interface has registered with the secondary MGC, the MG interface can automatically switch back to the primary MGC when the primary MGC recovers.	0: indicates that dual homing is not supported.
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	0: indicates that a wildcard is used.
6	Indicates whether the MG interface supports device authentication. To configure an MG interface to or not to support authentication, use this	1: indicates that device authentication is not supported.

Parameter	Description	Default Setting
	parameter. After the device authentication is supported, run the **auth(h248)** command to configure the authentication parameters, and then run the **reset(h248)** command to reset the MG interface. In this way, the MGC can manage the security of the MGs and avoid illegal registration with the MGC.	
7	Indicates whether the MG interface supports security header. To configure an MG interface to or not to support security header, use this parameter.	1: indicates that security header is not supported.
11	Indicates whether the MG interface supports emergency standalone. To configure whether an MG interface supports emergency standalone, use this parameter. If the MG interface supports emergency standalone, the users on the MG interface can make phone calls even if the MG fails to communicate with the MGC.	0: indicates that no call is permitted.
13	Digitmap matching mode	2: indicates the minimum matching.
15	Indicates whether the function of filtering media streams by source port is enabled on an MG interface. To enable or disable the function of filtering media streams by source port on an MG interface, use this parameter. When the function of filtering media streams by source port is enabled on the MG interface, only the media streams from the specific ports can be received.	0: indicates that media streams are not filtered by source port.

Parameter	Description	Default Setting
16	Indicates the length of the timer for filtering the media stream source port of the MG interface. To configure the length of the timer for filtering the media stream source port of an MG interface, use this parameter. When an MG interface does not filter the source port, the MG interface automatically filters the source port if the filtering timer times out.	5s
22	Indicates the type of the prompt tone that is played after the communication between the MG and the MGC is interrupted. To configure the type of the prompt tone after the communication between the MG and the MGC is interrupted, use this parameter.	0: indicates the busy tone.
23	Indicates the length of the timer for synchronizing the port status. To configure the length of the timer for synchronizing the port status, use this parameter.	35s
24	Indicates the maximum value of the Real-Time Transport Protocol (RTP) termination ID.	-
25	Indicates the maximum random value for the protection against avalanche of the H.248 interface.	-
26	Indicates the type of local blocking play tone.	0: indicates the busy tone.
27	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
28	Indicates the duration of the howler tone.	60s
29	Indicates the duration of message waiting tone.	3s

Parameter	Description	Default Setting
30	Indicates the time limit of the alarm for extra long call.	60 minutes
31	Indicates whether to report the alarm for extra long call.	1: indicates that the alarm is not reported.
32	Min. auto registration interval of remotely-blocked port(s).	1800s
33	Whether MG heartbeat is shut down.	1: No, heartbeat is enabled

There are 17 software parameters (numbered from 0 to 16) of an MG interface that supports MGCP. Table 1-42 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-42 Software parameters of an MG interface that supports MGCP

Parameter	Description	Default Setting
1	Indicates whether the ongoing call is maintained if the communication between the MG interface and the MGC is abnormal. To maintain the ongoing call when the communication between the MG interface and the MGC is abnormal, use this parameter.	1: disconnects all the calls at once.
2	Indicates whether the MG interface supports dual homing. To configure whether an MG interface supports dual homing, use this parameter. If the MG interface does not support dual homing (value: 1), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC.	0: indicates that dual homing is supported.
3	Indicates whether the heartbeat message between the MG and the MGC is disabled. To configure whether the	1: indicates that the heartbeat message is not disabled.

Parameter	Description	Default Setting
	heartbeat message between the MG and the MGC is disabled, use this parameter.	
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	0: indicates that a wildcard is used.
5	Indicates the MGC type. To select the MGC of a different type, use this parameter.	0
6	Indicates the maximum time threshold for responding to the heartbeat messages. To configure the maximum times for transmitting the heartbeat message continuously, use this parameter.	3
7	Indicates whether to report the heartbeat with the MG as an endpoint. To configure whether to report the heartbeat with the MG as an endpoint, use this parameter.	0: indicates that reporting the heartbeat with the MG as an endpoint is not supported.
10	Indicates the point-to-point (P2P) fault reporting. To configure whether the MG interface supports the P2P fault reporting from the ISDN terminals, use this parameter.	0: indicates that the P2P fault is reported.

Parameter	Description	Default Setting
11	Indicates the point-to-multipoint (P2MP) fault reporting. To configure whether the MG interface supports the P2MP fault reporting from the ISDN terminals, use this parameter.	1: indicates that the P2MP fault is not reported.
12	Indicates the type of local blocking play tone.	0: indicates the busy tone.
13	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
14	Indicates the RTP filtering switch of the MGCP interface. To configure whether the RTP filtering function is enabled, use this parameter. When the RTP filtering function is enabled, only the media stream from the specific ports can be received.	1: indicates that the RTP filtering function is not enabled.
15	Indicates the duration of the howler tone.	60s
16	Whether the timer symbol "T" follows the number string reported by the signaling.	0: Yes

Procedure

- When the system protocol is H.248, perform the following operations to configure the software parameters of an MG interface.
  a. In the global config mode, run the **interface h248** command to enter the MG interface mode.
  b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.
  c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.
- When the system protocol is MGCP, perform the following operations to configure the software parameters of an MG interface.
  a. In the global config mode, run the **interface mgcp** command to enter the MG interface mode.
  b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.
  c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 11 of H.248-based MG interface 0 to 1 so that the MG interface supports emergency standalone and allows only internal calls, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-software parameter 11 1
huawei(config-if-h248-0)#display mg-software parameter 11
  -------------------------------------------------
  Interface Id:0         para index:11  value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   11: Stand alone flag
       0: None
       1: Inner
       2: Emergency
       3: Both
```

1.22.1.5 (Optional) Configuring the TID Format of an MG Interface

The TID format determines how various user terminal IDs on the MG interface are generated.

Prerequisites

NOTICE

The configuration of the TID format is relatively complicated. The information such as the syntax rules with which the terminal prefix must comply and the requirements for the character string of the TID template is defined in the protocol, and is not described here. This topic describes only some basic information. It is recommended that you refer to the TID description in ITU-T H248.1 (applicable to H.248) or RFC 3435 (applicable to MGCP) before configuring the TID format.

Context

The TID format consists of the terminal prefix and the TID template. The TID template defines the generation mode of the TID excluding the terminal prefix. A TID consists of a terminal prefix and a character string generated by a TID template.

The TID templates that are bound to various types of users on the MG interface determine whether the users support terminal layering.

- If the parameter list of the TID template includes only keyword "G", the TID template is used by the non-layering users. Users bound with this template do not support terminal layering.
- If the parameter list of the TID template includes only keywords "F", "S", "P", "B" ("B" is not available to PSTN users), the TID template is used by the layering users. Users bound with this template support terminal layering.

NOTE

The meaning of each keyword is as follows:

- F indicates the subrack ID.
- S indicates the slot ID.
- P indicates the port ID.
- B indicates the B channel ID (only for ISDN terminals).
- G indicates the general permanent termination index.
- R indicates the RTP ephemeral termination index (only for the RTP ephemeral termination, which exists only when the system protocol is H.248. This termination is not involved unless special remarks are provided.)

When adding a user that supports terminal layering, you cannot and do not have to specify the parameter **terminalid** because the system automatically allocates a terminal ID. When adding a user that does not support terminal layering, you must specify the parameter **terminalid**.

You can run the **display tid-format(h248)** command or **display tid-format(mgcp)** command to query the TID formats of various types of users on the current MG interface. In the query result, **template-index** indicates the index of the TID template that is bound to the type of users. Then, run the **display tid-template** command to check whether the TID template supports the layering configuration. Hence, you can check whether the user supports terminal layering.

Precaution

- There are 19 default TID templates (templates 0-18) in the system. The default TID templates can be referenced, but cannot be added, modified, or deleted.
- The configuration of terminal layering on the MG must be the same as that on the MGC.
- If certain type of terminals exists on an interface and the interface is not disabled, the terminal prefix of this type of terminals cannot be modified.
- If a certain type of terminals exists on an interface, the TID format (including the terminal prefix and the index of the TID template) of this type of terminals cannot be changed.
- The terminal prefix must comply with the following syntax rules: The prefix must not exceed 64 characters. Only letters, digits, "_", and "/" are the characters allowed for input. The first character must be a letter, and the last character must not be a digit.
- The length of the TID, which is generated by combining the TID template and the terminal prefix that you configured, must not exceed 64 characters.

Procedure

- If the system protocol is H.248, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface h248** command to enter the H.248 mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).

NOTICE

The terminal prefix of PSTN, BRA and PRA must be all identical or unique.

    - In the H.248 mode, run the **tid-format rtp** command to configure the TID template and the terminal prefix of the RTP ephemeral termination.
    - In the H.248 mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
    - In the H.248 mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
    - In the H.248 mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.

  e. Run the **display tid-format(h248)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.

- If the system protocol is MGCP, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface mgcp** command to enter the MGCP mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).
    - In the MGCP mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
    - In the MGCP mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
    - In the MGCP mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.
  e. Run the **display tid-format(mgcp)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.

----End

Example

Assume that in the H.248 mode, the terminal prefix of the PSTN user on MG interface 1 is aln/, and the layering TID template 3 is used. To add a PSTN user on port 0/2/0 and check whether the system automatically allocates a TID generated according to the template, do as follows:

```
huawei(config)#display tid-template 3//Query the information about TID template 3
  -------------------------------------------------
  Index     : 3
  Format    : %u/%u/%u
  Para-list  : F+1,S+1,P+1  //The parameter list of the TID template includes keyword
"F", "S",
                //and "P", which indicates that this template supports terminal layering.
  Name      : Aln_Not_Fixed_1
  -------------------------------------------------
huawei(config)#interface h248 1
huawei(config-if-h248-1)#tid-format pstn prefix aln template 3
huawei(config-if-h248-1)#quit
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser add 0/2/0 1
huawei(config-esl-user)#display mgpstnuser 0/2/0
{ <cr>|endframeid/slotid/portid<S><1,15> }:

  Command:
         display mgpstnuser 0/15/0
  ------------------------------------------------------------------------
  F  /S /P    MGID    TelNo          Priority PotsLineType TerminalID
  ------------------------------------------------------------------------
  0  /2 /0    1       -              Cat3     DEL          aln/1/3/1
           //The system allocates the terminal ID according to the TID format.
  ------------------------------------------------------------------------
```

1.22.1.6 Enabling an MG Interface

Enabling an MG interface is to reset an MG interface to make the MG interface register with the MGC (or to make the modified attributes of the MG interface take effect) after the configuration of the MG interface is complete, so that the MG interface can work in the normal state.

Precaution

⚠ CAUTION

For the MG interface that has been in service, this operation interrupts the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation.

Procedure

- Enable the MG interface that adopts the H.248 protocol.
  a. Run the **interface h248** command to enter the H.248 mode.
  b. Run the **reset coldstart** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-h248 all** command to check whether the MG interface is in the normal state.
- Enable the MG interface that adopts the MGCP protocol.
  a. Run the **interface mgcp** command to enter the MGCP mode.
  b. Run the **reset** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-mgcp all** command to check whether the MG interface is in the normal state.

----End

Example

To enable H.248-based MG interface 0, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-h248-0)#quit
huawei(config)#display if-h248 all
--------------------------------------------------------------------------------
MGID    Trans    State        MGPort MGIP         MGCPort MGCIP/DomainName
--------------------------------------------------------------------------------
0       UDP      Normal        2944   10.10.10.11     2944   10.10.20.11
--------------------------------------------------------------------------------
```

To enable MGCP-based MG interface 0, do as follows:

```
huawei(config)#interface mgcp 0
huawei(config-if-mgcp-0)#reset
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-mgcp-0)#quit
huawei(config)#display if-mgcp all
  ------------------------------------------------------------------------
  MGID    State         MGPort MGIP          MGCPort MGCIP/DomainName
  ------------------------------------------------------------------------
  0       Normal        2727   10.10.10.11     2727    10.10.20.11
  1       Wait ack      2527   10.10.10.12     2727    10.10.20.12
  ------------------------------------------------------------------------
```

1.22.2 Configuring the IUA Link

This topic describes how to configure the IUA link for signaling transmission between the Access node and MGC in the VoIP ISDN service.

Context

- Simple Control Transmission Protocol (SCTP) is a connection-oriented protocol. Its most fundamental function is to provide reliable transmission for interaction messages between the Access node and MGC. The SCTP protocol implements services based on the association between two SCTP endpoints. SCTP can be regarded as a transmission layer. Its upper layer is called SCTP subscriber, and its lower layer is the IP network.
- The IUA link is the carrier of the interaction signaling between the Access node and MGC.

1.22.2.1 Adding an IUA Link Set

This topic describes how to add an IUA link set. When configuring the VoIP ISDN service, you need to configure the IUA link to carry the Q.931 call signaling. Before adding an IUA link, you must add a corresponding link set. Otherwise, the link cannot be added.

Context

- The system supports the configuration of a maximum of IUA link sets.
- After a link set is configured successfully, it is in the deactivated state by default.

Procedure

Step 1 In global config mode, run the command **sigtran** to enter Sigtran mode.

Step 2 Run the **iua-linkset add** command to add an IUA link set.

Step 3 You can run the **display iua-linkset attribute** command to check whether the configured IUA link set information is the same as the data plan.

----End

Example

Assume that the link set ID is 0, working mode of the link set is active/standby mode, pending duration is 20s, prefix of the IID is b/, IID generation mode is using the binary value that is automatically generated in ffsspp mode, namely, parameter 2. To add such an IUA link set, do as follows:

```
huawei(config)#sigtran
huawei(config-sigtran)#iua-linkset add 0 trafficmode override pendingtime 20 iid-map
2
braprefix b/
huawei(config-sigtran)#display iua-linkset attribute
{ <cr>|linksetno<L> }:

  Command:
         display iua-linkset attribute
-----------------------------------------------------------
  LinksetNo           :0
  PendingTime          :20
  TrafficMode          :override
  C/S-Mode            :server
  IID-Type            :integer
  IID-Map             :2
  BRA IID-Prefix       :b/
  BRA IID-Suffix       :-
-----------------------------------------------------------
```

1.22.2.2 Adding an IUA Link

This topic describes how to add an IUA link. After the link set is added, you can add an IUA link to carry the Q.931 call signaling for the ISDN user.

Prerequisites

The IUA link set must be added.

Context

- Make sure that a minimum of one item in the local port ID, local IP address, remote port ID, and remote IP address of a link is different from the corresponding item of other links.
- Only two links can be configured in the same link set. In addition, the local IP addresses of the two links must be the same.

Procedure

Step 1 (This step is not required if the command line interface is already in the Sigtran mode.) In global config mode, run the **sigtran** command to enter the Sigtran mode.

Step 2 Run the **iua-link add** command to add an IUA link.

Step 3 You can run the **display iua-link attribute** command to check whether the configured IUA link information is the same as the data plan.

----End

Example

Assume that the link ID is 0, link set ID is 0, local port ID is 1402, local IP address is 10.10.10.10, remote port ID is 1404, and remote IP address 1 is 10.10.10.20. To add such a link, do as follows:

```
huawei(config)#sigtran
huawei(config-sigtran)#iua-link add 0 0 1402 10.10.10.10 1404 10.10.10.20
huawei(config-sigtran)#display iua-link attribute
{ <cr>|linkno<L> }:

  Command:
        display iua-link attribute
  -----------------------------------------------------------------------
  LinkNo              :  0
  LinksetNo           :  0
  Local port           :  1402
  Local IP address      :  10.10.10.10
  Remote port          :  1404
  Remote IP address     :  10.10.10.20
  Remote IP address 2   :  -
  Priority            :  0
  -----------------------------------------------------------------------
```

1.22.3 Configuring the VoIP ISDN BRA User

After the MG interface is configured, you can add the VoIP ISDN BRA user on this interface, and configure the system parameters, oversea parameters, and attributes of the BRA port, to implement the VoIP ISDN BRA service.

1.22.3.1 Configuring the ISDN BRA User Data

This topic describes how to configure the ISDN BRA user data based on H.248 (including the priority, flag of reporting the fault of UNI, etc., and the data must be the same as the corresponding data on the MGC) so that the ISDN BRA user can access the network to use the ISDN BRA service.

Prerequisites

The ISDN BRA service board must be inserted into the planned slot correctly and the board must be in the normal state.

The IUA link must be configured according to the requirements. For details about how to configure the IUA link, see 1.22.2 Configuring the IUA Link.

Default Configuration

Table 1-43 lists the default settings of the attributes of the ISDN BRA user. When configuring these attributes, you can modify the values according to the service requirements.

Table 1-43 Default settings of the attributes of the ISDN BRA user

Parameter	Default Settings
Priority of the ISDN BRA user	cat3: (common user)
Flag of reporting the UNI fault of the ISDN BRA user	Disable
Threshold for the number of auto recoveries from deterioration faults	20

Procedure

Step 1 In global config mode, run the **esl user** command to enter the ESL user mode.

Step 2 Run the **mgbrauser add** command to add an ISDN BRA.

- When **iid-map** in the **iua-linkset add** command is configured to 1, interfaceid must be configured and be different from the interfaceid of other users in the same link set.
- The terminal ID of an ISDN BRA user must be different from the terminal IDs of other users.
- If the MG interface does not support the terminal layering function, the terminal ID must be configured when an ISDN BRA user is added. In addition, the terminal ID must differ from the terminal ID of the existing ISDN BRA user by an integer multiple of 2. For example, to add the first ISDN BRA user, the terminal ID is 2; to add the second ISDN BRA user, the terminal ID is 4; to add the third ISDN BRA user, the terminal ID is 6; the rest may be deduced by analogy.
- If the MG interface supports the terminal layering function, the terminal ID cannot be configured when an ISDN BRA user is added on the MG interface. The system automatically allocates a terminal ID for the user.

Step 3 Run the **mgbrauser attribute set** command to configure the attributes of the ISDN BRA user.

The attributes of an ISDN BRA user include the following:

- Priority of the ISDN BRA user. The priorities are classified to cat1 (the first class government user), cat2 (the second class government user), and cat3 (common user) in the sequence of descending. Without special requirements, the default cat3 is adopted.
- Flag of reporting the UNI fault of the ISDN BRA user. This attribute determines whether to report the UNI fault to MGC. The default is not to report.
- Threshold for the number of auto recoveries from deterioration faults. This attribute indicates the maximum times that the equipment attempts to recover from deterioration faults. Zero indicates not to recover automatically. The number of 255 indicates not to limit the attempt times. The default is 20.

Step 4 Run the **display mgbrauser attribute** command to query whether the configured attributes of the ISDN BRA user are the same as the data plan.

----End

Example

Assume the following configurations:

- Link set ID: 0
- IUA interface ID: 0 (value of **iid-map**: 1)
- Terminal ID: 2 (not supporting the terminal layering function)
- User priority: cat3
- Telephone number: 83110001 (Before configuring a emergency standalone number, ensure that the emergency standalone function has been enabled on the MG interface. For details, see 1.22.1.4 (Optional) Configuring the Software Parameters of an MG Interface.)
- UNI alarm report function: enable
- Threshold for the number of auto recoveries from L1 deterioration faults: 30

To add an ISDN BRA user with such configurations on port 0/4/0 of MG interface 0, do as follows:

```
huawei(config-sigtran)#quit
huawei(config)#esl user
huawei(config-esl-user)#mgbrauser add 0/4/0 0 0 interfaceid 0 terminalid 2 priority
cat3 telno 83110001
 Are you sure to configure the working mode of the DSL board to normal and reset
 the board automatically? (y/n)[n]:y
huawei(config-esl-user)#display mgbrauser 0/4
  ------------------------------------------------------------------------------
  F  /S /P /B  MGID    LinkSetNo UserIFID TelNo          Priority TerminalID
  ------------------------------------------------------------------------------
  0/4/0 /0  0       0        0       83110001        Cat3     A2
  ------------------------------------------------------------------------------
huawei(config-esl-user)#mgbrauser attribute set 0/4/0 priority cat3 unireport enable
auto-resume-limit 30
huawei(config-esl-user)#display mgbrauser attribute 0/4/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
         display mgbrauser attribute 0/4/0
 --------------------------------------------------------------------------
  F  /S /P                      :  0/4/0
  UNIreport                     :  enable
  Priority                      :  Cat3
  Auto reservice times/limit        :  0/30
  DSP-para-template                 :  -
  --------------------------------------------------------------------------
```

1.22.3.2 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  ------------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
  ------------------------------------------------------------------------------
```

1.22.3.3 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1

  Command:
         display oversea parameters 1
    ------------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 100
```

```
Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
-----------------------------------------------------------------------------
```

1.22.3.4 (Optional) Configuring the Attributes of an ISDN BRA Port

This topic describes how to configure the attributes of an ISDN BRA port to ensure that the ISDN BRA port can meet the actual application requirements. You can configure the auto-deactivation status, remote power supply status, UNI fault alarming function, and working mode of the port.

Default Configuration

Table 1-44 lists the default values of the attributes of an ISDN BRA port. When configuring these attributes, you can modify the values according to the service requirements.

Table 1-44 Default values of the attributes of an ISDN BRA port

Parameter	Default Setting
Autodeactive	Disable
Autodeactive-delay	30s
Activemode	unstable-active
Remotepower	Disable
Unialarm	Disable
Workmode	p2mp

Procedure

Step 1 In global config mode, run the **braport** command to enter braport mode.

Step 2 Run the **braport attribute set** command to configure the attributes such as the working mode, auto-deactivation status, and remote power supply status of the port.

If an ISDN BRA port needs to be connected to multiple terminal users, configure the working mode of the port to p2mp. If an ISDN BRA port needs to be connected to only one terminal user, configure the working mode of the port to p2p.

For detailed description of the **braport attribute set** command, see the parameter description in **braport attribute set**.

----End

Example

Assume that the working mode is p2mp, the activation mode is stable, and the auto-deactivation function is disabled. To configure such attributes of ISDN BRA port 0/4/0, do as follows:

```
huawei(config)#braport
huawei(config-braport)#braport attribute set 0/4/0 workmode p2mp activemode
```

```
stable-active
huawei(config-braport)#display braport attribute
{ frameid/slotid/portid<S><Length 1-15>|frameid/slotid<S><Length 1-15> }:0/4/0

  Command:
          display braport attribute 0/4/0
-----------------------------------------------------------------------------
F  /S /P  Remotepower Workmode Autodeactive Deactivedelay Activemode Unialarm
-----------------------------------------------------------------------------
0/4/0  disable    p2mp     disable     30          stable    disable
-----------------------------------------------------------------------------
```

1.23 Configuring the VoIP ISDN PRA Service (H.248-based)

This topic describes how to configure the VoIP ISDN PRA service on an IP network. When the Access node uses the H.248 protocol, the device supports the access of the ISDN PRA user. ISDN provides end-to-end (E2E) digital connection and supports multiple types of voice and non-voice telecom services.

Prerequisites

According to the actual network, a route from the Access node to the MGC must be configured to ensure that the Access node communicates with the MGC normally.

Context

- The voice over IP (VoIP) service uses the IP packet switched network for transmission after the traditional analog voice signals are compressed and packetized. This service lowers the cost of the voice service. For the detailed description of the VoIP service, see Voice Feature in the *Feature Description*.
- Defined by the International Telegraph and Telephone Consultative Committee (CCITT), the ISDN is a communication network evolved from the Integrated Digital Network (IDN). The ISDN service provides the E2E digital connection and supports multiple types of voice and non-voice telecom services. On the ISDN network, users can access the network through the following two interfaces: (The ISDN technology provides two types of user-network interfaces based on the user-network interface reference model.)
  - ISDN basic rate interface (BRI), which supports a rate of 144 kbit/s and provides two B channels (for carrying services) and one D channel (for transmitting call control signaling and maintenance and management signaling). The rate of B channels is 64 kbit/s and the rate of D channel is 16 kbit/s. The service carried on the ISDN BRI is the ISDN basic rate access (BRA) service.
  - ISDN primary rate interface (PRI), which supports a rate of 2.048 Mbit/s and provides 30 B channels and one D channel. The rates of the B channel and D channel are both 64 kbit/s. The service carried on the ISDN PRI is the ISDN primary rate access (PRA) service.

Precaution

The media gateway control protocol (MGCP) is a master/slave protocol, under which the MGC controls the AG to implement the call connection. The data on the AG for the interconnection with the MGC, such as the attributes of the MG interface and the attributes of the VoIP user, must be the same as the corresponding data on the MGC. Therefore, before configuring the VoIP service, you must contact MGC engineers to check and ensure that the interconnection data plan for the AG is consistent with the corresponding plan for the MGC.

Data preparation

Table 1-45 provides the data plan for configuring the H.248-based VoIP ISDN PRA service.

Table 1-45 Data plan for configuring the H.248-based VoIP ISDN PRA service

Item			Remarks
MG interface data (The data must be consistent with the data on the MGC.)	Parameters of the media stream and signaling stream	Upstream VLAN for media and signaling streams	It is used for the upstream VLAN of the VoIP service to be configured. Standard VLAN is recommended.
		Uplink port for media and signaling streams	It is used as the uplink port for the VoIP service to be configured.
		Media IP address and signaling IP address	These IP addresses must be contained in the media and signaling IP address pools. The media and signaling IP address pools consist of all the IP addresses of the Layer 3 interface of the upstream VLAN for media and signaling streams.
		Default IP address of the MG	It is the next hop IP address from the Access node to the MGC.
	Attribute parameters of the MG interface **NOTE** Parameters listed here are mandatory, which means that the MG interface cannot be enabled if these parameters are not configured.	MG interface ID	It is the ID of the MG interface used by the VoIP service to be configured. The Access node supports only one VAG.
		Signaling port ID of the MG interface	It is the transport layer protocol port ID used for the signaling exchange between the Access node (AG) and the MGC. The default signaling port ID defined in H.248 is 2944 (text) and 2945 (binary).
		IP address of the primary MGC to which the MG interface belongs	When dual homing is not configured, you can configure the parameters of only the primary MGC. When dual homing is configured, you also need to configure the IP address

Item			Remarks
		Port ID of the primary MGC to which the MG interface belongs	and port ID of the secondary MGC.
		Coding mode of the MG interface	Currently, only the **text** mode is supported.
		Transmission mode of the MG interface	The transmission mode is selected according to the requirements of the MGC. Generally, UDP is used.
		Domain name of the MG interface	It corresponds to the **domainName** parameter of the MG interface. When the H.248 protocol is used, this parameter must be configured if the **MIDType** parameter of the H.248 message is configured to **domainName**. Otherwise, the MG interface cannot be started. In other situations, this parameter is optional.
		Device name of the MG interface	It is supported by only the H.248 protocol, and it corresponds to the **deviceName** parameter of the MG interface that uses the H.248 protocol. This parameter must be configured if the **MIDType** parameter of the H.248 message is configured to **domainName**. Otherwise, the MG interface cannot be started. In other situations, this parameter is optional.
	Software parameters of the MG interface		Whether the MG interface supports the functions such as dual homing and emergency standalone is determined by the service requirements.
	Terminal ID (TID) format of the MG interface		The TID format determines the generation mode of various types of user terminals on an MG interface.
IUA link	IUA link set		The IUA link can be configured

Item			Remarks
			only after the IUA link set is configured.
	IUA link **NOTE** The local port ID, local IP address, remote port ID, and remote IP address of different links must not be completely same; otherwise, the service cannot be configured.	IUA link ID	It indicates the link for transmitting the signaling.
		IUA link set ID	-
		Local port ID	To activate the link normally, it must be the same as the remote port ID configured on the MGC.
		Local IP address	It must be the same as the remote IP address of the link configured on the MGC. In addition, the local IP addresses of the links that are in the same link set must be the same. (The IP address must exist in the media IP address pool.)
		Remote port ID	To activate the link normally, it must be the same as the local port ID configured on the MGC.
		Remote IP address	It must be the same as the local IP address of the link configured on the MGC. The SCTP protocol supports the multi-homing function. That is, one link can be configured with the IP addresses of multiple MGCs as the remote IP addresses. When one MGC is faulty, the link can be switched to other MGCs automatically. This ensures that the service is not affected. The Access node supports the configuration of the active remote IP address and standby remote IP address.
ISDN PRA user data (The data must be consistent with the data on the MGC.)	Slot that houses the E1 service board		-
	User data	Termination ID	If the TID format bound to the PRA user does not support terminal layering function, this parameter needs to be configured, and the configuration must be consistent with the configuration on the MGC.

Item			Remarks
		User priority	The user priority must be specified according to the service requirements. There are three categories of user priorities, which are as follows: • cat1: government1 (category 1 government user) • cat2: government2 (category 2 government user) • cat3: normal (common user, default)
		Interface ID	It indicates the interface for the PRA user data to pass through the MG and MGC. The configuration of this parameter must be consistent with the corresponding configuration on the MGC.
	System parameters		The system parameters including the oversea version flag and message waiting indication (MWI) mode need to be configured according to local standards to ensure that the response of the user terminal complies with local standards.
	Overseas parameters		The attributes such as the upper and lower thresholds of the flash-hooking duration need to be configured according to local standards to ensure that the response of the user terminal complies with local standards.
	E1 port attributes		The attributes include the access mode of the board, port mode, line coding mode, and port impedance. Generally, if there is no requirement, these attributes need not be modified.

1.23.1 Configuring an MG Interface

This topic describes how to configure an MG interface for implementing the communication between the MA5600T/MA5603T/MA5608T (AG) and the MGC.

Context

- The MA5600T/MA5603T/MA5608T communicates with the MGC through either the H.248 or the MGCP protocol. One MA5600T/MA5603T/MA5608T can run only one protocol.
- One MA5600T/MA5603T/MA5608T supports up to eight MG interfaces. If a CKMC daughter board is configured in the MA5600T/MA5603T/MA5608T, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently. Each MG interface can be configured with the attributes (such as authentication and ringing mapping) independently.

Procedure

1.23.1.1 Configuring the Upstream VLAN Interface

This topic describes how to specify the upstream VLAN interface for the media stream and the signaling flow, and how to configure the IP addresses of the Layer 3 interface. These IP addresses are the sources of the media and signaling IP address pools.

Context

Multiple IP addresses can be configured for the same VLAN Layer 3 interface. Only one IP address functions as the primary address, and other IP addresses function as the secondary addresses.

Procedure

Step 1 Run the **vlan** command to add an upstream VLAN for the media stream and the signaling flow.

Step 2 Run the **port vlan** command to add the upstream ports to the VLAN.

Step 3 Configure the IP addresses of the VLAN Layer 3 interface.

1. Run the **interface vlanif** command to enter the Layer 3 interface of the upstream VLAN for the media stream and the signaling flow.
2. Run the **ip address** command to configure the IP addresses of the Layer 3 interface.

Step 4 Run the **display interface vlanif** command to check whether the IP addresses of the Layer 3 interface are the same as those in the data plan.

----End

Example

Assume that the media stream and the signaling stream are transmitted upstream through upstream port 0/19/0. To create media and signaling upstream VLAN 3 and configure IP addresses 10.13.4.116/16 and 10.13.4.117/16 of the Layer 3 interfaces for the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#vlan 3 standard
huawei(config)#port vlan 3 0/19 0
huawei(config)#interface vlanif 3
huawei(config-if-vlanif3)#ip address 10.13.4.116 16
huawei(config-if-vlanif3)#ip address 10.13.4.117 16 sub
huawei(config-if-vlanif3)#quit
huawei(config)#display interface vlanif 3
vlanif3 current state : UP
Line protocol current state : UP
Description : HUAWEI, SmartAX Series, vlanif3 Interface
The Maximum Transmit Unit is 1500 bytes
Forward plane MTU: -
Internet Address is 10.13.4.116/16
Internet Address is 10.13.4.117/16 Secondary
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 00e0-fc32-1118
VLAN Encap-mode : single-tag
```

1.23.1.2 Configuring the Media and Signaling IP Address Pools

The media IP address pool defines all media IP addresses that can be used by the AG, and the signaling IP address pool defines all signaling IP addresses that can be used by the AG.

Prerequisites

The IP address of the Layer 3 interface of the media and signaling upstream VLAN must be configured. For details about how to configure the IP address, see 1.23.1.1 Configuring the Upstream VLAN Interface.

Context

- The media IP address and the signaling IP address for the MG interface must be selected from the IP address pools configured here.
- The signaling IP address pool is used to store the IP addresses of the MG interfaces, and the media IP address pool is used to store the IP addresses of the media streams controlled by the signaling.
- The media IP address pool and the signaling IP address pool can be the same. Similarly, the media IP address and the signaling IP address can be the same.

NOTICE

The MGCP interface on the MA5600T/MA5603T/MA5608T does not support the isolation of media streams and signaling flows. Therefore, when the MGCP protocol is used, the media IP address pool and the signaling IP address pool must be the same. When the H.248 or SIP protocol is used, the media IP address pool and the signaling IP address pool can be the same or different.

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Configure the media IP address pool.

1. Run the **ip address media** command to add the media IP address to the media IP address pool.

   The media IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.

2. Run the **display ip address media** command to check whether the media IP address pool is the same as that in the data plan.

Step 3 Configure the signaling IP address pool.

1. Run the **ip address signaling** command to add the signaling IP address to the signaling IP address pool.

   The signaling IP address needs to be selected from the IP addresses of the Layer 3 interface of the media and signaling upstream VLAN.
2. Run the **display ip address signaling** command to check whether the signaling IP address pool is the same as that in the data plan.

----End

Example

To add IP addresses 10.13.4.116/16 and 10.13.4.117/16 of Layer 3 interfaces of the media and signaling upstream VLAN to the media IP address pool and the signaling IP address pool respectively, do as follows:

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.13.4.116 10.13.0.1
huawei(config-voip)#ip address media 10.13.4.117 10.13.0.1
huawei(config-voip)#ip address signaling 10.13.4.116
huawei(config-voip)#ip address signaling 10.13.4.117
huawei(config-voip)#display ip address media
  Media:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  Gateway..............: 10.13.0.1
  MAC Address..........: 00-E0-FC-AF-91-33
huawei(config-voip)#display ip address signaling
  Signaling:
  IP Address...........: 10.13.4.116
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33

  IP Address...........: 10.13.4.117
  Subnet Mask..........: 255.255.0.0
  MAC Address..........: 00-E0-FC-AF-91-33
```

1.23.1.3 Adding an MG Interface

This topic describes how to add an MG interface, through which the Access node can communicate with the MGC.

Context

- One Access node supports a maximum of 8 MG interfaces. If a CKMC daughter board is configured in the Access node, up to 32 MG interfaces can be supported. Each MG interface can be configured with the interface attributes independently.

- The configuration of the attributes of an MG interface is valid only to the MG interface.

Procedure

- Add an MG interface that supports H.248.
  a. Run the **display protocol support** command to query the current system protocol.
     - If the current system protocol is H.248, go to h.
     - If the current system protocol is MGCP, go to b.

  b. Run the **display if-mgcp all** command to query whether an MG interface that supports MGCP exists.
     - If such an MG interface does not exist, go to e.
     - If such an MG interface exists, go to c.

  c. Delete all configuration data of this MG interface, and then run the **shutdown(mgcp)** command to disable the MG interface.

NOTICE

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

  d. Run the **undo interface mgcp** command to delete the MG interface.

  e. Run the **protocol support** command to change the system protocol to H.248.

  f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.

  g. After the system is restarted, log in to the system, and enter the global config mode.

  h. Run the **interface h248** command to add an MG interface that supports H.248.

  i. Run the **if-h248 attribute** command to configure the attributes of the MG interface according to the data plan.

  j. Run the **display if-h248 attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

- Add an MG interface that supports MGCP.
  a. Run the **display protocol support** command to query the current system protocol.
     - If the current system protocol is MGCP, go to h.
     - If the current system protocol is H.248, go to b.

  b. Run the **display if-h248 all** command to query whether an MG interface that supports H.248 exists.
     - If such an MG interface does not exist, go to e.
     - If such an MG interface exists, go to c.

  c. Delete all configuration data of this MG interface, and then run the **shutdown(h248)** command to disable the MG interface.

⚠CAUTION

This operation directly interrupts all the services on the MG interface. Hence, exercise caution when performing this operation.

d. Run the **undo interface h248** command to delete the MG interface.

e. Run the **protocol support** command to change the system protocol to MGCP.

f. Run the **save** command to save the configuration data, and then run the **reboot system** command to restart the system to make the new configuration data take effect.

g. After the system is restarted, log in to the system, and enter the global config mode.

h. Run the **interface mgcp** command to add an MG interface that supports MGCP.

i. Run the **if-mgcp attribute** command to configure the attributes of the MG interface according to the data plan.

j. Run the **display if-mgcp attribute** command to check whether the attributes of the MG interface are the same as those in the data plan.

----End

Example

Assume that the MG interface ID is 0 and the H.248 protocol is used for interconnecting with the MGC. To add the MG interface, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#interface h248 0
  Are you sure to add MG interface?(y/n)[n]:y
```

Assume that the MG interface ID is 0 and the MGCP protocol is used for interconnecting with the MGC. The current system protocol, however, is the H.248 protocol. It is confirmed that the H.248 interface exists in the system but is not in use. To add MG interface 0, do as follows:

```
huawei(config)#display protocol support
System support H248 protocol
huawei(config)#display if-h248 all
  ----------------------------------------------------------------------------
  MGID     Trans     State     MGPort MGIP          MGCPort MGCIP/DomainName
  ----------------------------------------------------------------------------
  0       -        Closed    -     -             -      -
  ----------------------------------------------------------------------------
huawei(config)#undo interface h248 0
  Are you sure to del MG interface?(y/n)[n]:y
huawei(config)#protocol support mgcp
huawei(config)#save
huawei(config)#reboot system
  Please check whether data has saved, the unsaved data will lose if reboot
system, are you sure to reboot system? (y/n)[n]:y
```

After the system is restarted, re-log in to the system.

```
huawei(config)#display protocol support
System support MGCP protocol
```

```
huawei(config)#interface mgcp 0
 Are you sure to add MG interface?(y/n)[n]:y
```

1.23.1.4 (Optional) Configuring the Software Parameters of an MG Interface

The software parameters of an MG interface mainly define certain common service attributes of the MG interface. After the configuration of the software parameters of an MG interface, the software parameters take effect immediately and the configuration is valid only to the MG interface.

Context

There are 34 software parameters (numbered from 0 to 33) of an MG interface that supports H.248. Table 1-46 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-46 Software parameters of an MG interface that supports H.248

Parameter	Description	Default Setting
2	Indicates whether the MG interface supports dual homing. To configure an MG interface to or not to support dual homing, use this parameter. If the MG interface does not support dual homing (value: 0), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC. If the MG interface supports dual homing and auto-switching (value: 2), even if the MG interface has registered with the secondary MGC, the MG interface can automatically switch back to the primary MGC when the primary MGC recovers.	0: indicates that dual homing is not supported.
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the	0: indicates that a wildcard is used.

Parameter	Description	Default Setting
	quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	
6	Indicates whether the MG interface supports device authentication. To configure an MG interface to or not to support authentication, use this parameter. After the device authentication is supported, run the **auth(h248)** command to configure the authentication parameters, and then run the **reset(h248)** command to reset the MG interface. In this way, the MGC can manage the security of the MGs and avoid illegal registration with the MGC.	1: indicates that device authentication is not supported.
7	Indicates whether the MG interface supports security header. To configure an MG interface to or not to support security header, use this parameter.	1: indicates that security header is not supported.
11	Indicates whether the MG interface supports emergency standalone. To configure whether an MG interface supports emergency standalone, use this parameter. If the MG interface supports emergency standalone, the users on the MG interface can make phone calls even if the MG fails to communicate with the MGC.	0: indicates that no call is permitted.
13	Digitmap matching mode	2: indicates the minimum matching.

Parameter	Description	Default Setting
15	Indicates whether the function of filtering media streams by source port is enabled on an MG interface. To enable or disable the function of filtering media streams by source port on an MG interface, use this parameter. When the function of filtering media streams by source port is enabled on the MG interface, only the media streams from the specific ports can be received.	0: indicates that media streams are not filtered by source port.
16	Indicates the length of the timer for filtering the media stream source port of the MG interface. To configure the length of the timer for filtering the media stream source port of an MG interface, use this parameter. When an MG interface does not filter the source port, the MG interface automatically filters the source port if the filtering timer times out.	5s
22	Indicates the type of the prompt tone that is played after the communication between the MG and the MGC is interrupted. To configure the type of the prompt tone after the communication between the MG and the MGC is interrupted, use this parameter.	0: indicates the busy tone.
23	Indicates the length of the timer for synchronizing the port status. To configure the length of the timer for synchronizing the port status, use this parameter.	35s
24	Indicates the maximum value of the Real-Time Transport Protocol (RTP) termination ID.	-

Parameter	Description	Default Setting
25	Indicates the maximum random value for the protection against avalanche of the H.248 interface.	-
26	Indicates the type of local blocking play tone.	0: indicates the busy tone.
27	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
28	Indicates the duration of the howler tone.	60s
29	Indicates the duration of message waiting tone.	3s
30	Indicates the time limit of the alarm for extra long call.	60 minutes
31	Indicates whether to report the alarm for extra long call.	1: indicates that the alarm is not reported.
32	Min. auto registration interval of remotely-blocked port(s).	1800s
33	Whether MG heartbeat is shut down.	1: No, heartbeat is enabled

There are 17 software parameters (numbered from 0 to 16) of an MG interface that supports MGCP. Table 1-47 lists the configurable parameters, and the other parameters are reserved in the system.

Table 1-47 Software parameters of an MG interface that supports MGCP

Parameter	Description	Default Setting
1	Indicates whether the ongoing call is maintained if the communication between the MG interface and the MGC is abnormal. To maintain the ongoing call when the communication between the MG interface and the MGC is abnormal, use this parameter.	1: disconnects all the calls at once.
2	Indicates whether the MG interface supports dual homing. To configure whether an MG interface supports dual	0: indicates that dual homing is supported.

Parameter	Description	Default Setting
	homing, use this parameter. If the MG interface does not support dual homing (value: 1), even if the secondary MGC is configured, the MG interface does not switch to register with the secondary MGC when the MG interface fails to register with the primary MGC.	
3	Indicates whether the heartbeat message between the MG and the MGC is disabled. To configure whether the heartbeat message between the MG and the MGC is disabled, use this parameter.	1: indicates that the heartbeat message is not disabled.
4	Indicates whether a wildcard is used for the registration of the MG interface. To configure whether a wildcard is used for the registration of an MG interface, use this parameter. Using a wildcard for registration can reduce the quantity of messages between the MG interface and the MGC. When the wildcard is not used for registration, all the terminals on the MG interface need to register with the MGC in order. The registration without a wildcard is also called "single-endpoint registration".	0: indicates that a wildcard is used.
5	Indicates the MGC type. To select the MGC of a different type, use this parameter.	0
6	Indicates the maximum time threshold for responding to the heartbeat messages. To configure the maximum times for transmitting the heartbeat message continuously, use this parameter.	3

Parameter	Description	Default Setting
7	Indicates whether to report the heartbeat with the MG as an endpoint. To configure whether to report the heartbeat with the MG as an endpoint, use this parameter.	0: indicates that reporting the heartbeat with the MG as an endpoint is not supported.
10	Indicates the point-to-point (P2P) fault reporting. To configure whether the MG interface supports the P2P fault reporting from the ISDN terminals, use this parameter.	0: indicates that the P2P fault is reported.
11	Indicates the point-to-multipoint (P2MP) fault reporting. To configure whether the MG interface supports the P2MP fault reporting from the ISDN terminals, use this parameter.	1: indicates that the P2MP fault is not reported.
12	Indicates the type of local blocking play tone.	0: indicates the busy tone.
13	Indicates the type of remote blocking play tone.	0: indicates the busy tone.
14	Indicates the RTP filtering switch of the MGCP interface. To configure whether the RTP filtering function is enabled, use this parameter. When the RTP filtering function is enabled, only the media stream from the specific ports can be received.	1: indicates that the RTP filtering function is not enabled.
15	Indicates the duration of the howler tone.	60s
16	Whether the timer symbol "T" follows the number string reported by the signaling.	0: Yes

Procedure

- When the system protocol is H.248, perform the following operations to configure the software parameters of an MG interface.

a. In the global config mode, run the **interface h248** command to enter the MG interface mode.

b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.

c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

- When the system protocol is MGCP, perform the following operations to configure the software parameters of an MG interface.

  a. In the global config mode, run the **interface mgcp** command to enter the MG interface mode.

  b. Run the **mg-software parameter** command to configure the software parameters listed in the data plan.

  c. Run the **display mg-software parameter** command to check whether the software parameters of the MG interface are the same as those in the data plan.

----End

Example

To configure software parameter 11 of H.248-based MG interface 0 to 1 so that the MG interface supports emergency standalone and allows only internal calls, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-software parameter 11 1
huawei(config-if-h248-0)#display mg-software parameter 11
  -------------------------------------------------
  Interface Id:0         para index:11  value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   11: Stand alone flag
      0: None
      1: Inner
      2: Emergency
      3: Both
```

1.23.1.5 (Optional) Configuring the TID Format of an MG Interface

The TID format determines how various user terminal IDs on the MG interface are generated.

Prerequisites

NOTICE

The configuration of the TID format is relatively complicated. The information such as the syntax rules with which the terminal prefix must comply and the requirements for the character string of the TID template is defined in the protocol, and is not described here. This topic describes only some basic information. It is recommended that you refer to the TID description in ITU-T H248.1 (applicable to H.248) or RFC 3435 (applicable to MGCP) before configuring the TID format.

Context

The TID format consists of the terminal prefix and the TID template. The TID template defines the generation mode of the TID excluding the terminal prefix. A TID consists of a terminal prefix and a character string generated by a TID template.

The TID templates that are bound to various types of users on the MG interface determine whether the users support terminal layering.

- If the parameter list of the TID template includes only keyword "G", the TID template is used by the non-layering users. Users bound with this template do not support terminal layering.
- If the parameter list of the TID template includes only keywords "F", "S", "P", "B" ("B" is not available to PSTN users), the TID template is used by the layering users. Users bound with this template support terminal layering.

NOTE

The meaning of each keyword is as follows:

- F indicates the subrack ID.
- S indicates the slot ID.
- P indicates the port ID.
- B indicates the B channel ID (only for ISDN terminals).
- G indicates the general permanent termination index.
- R indicates the RTP ephemeral termination index (only for the RTP ephemeral termination, which exists only when the system protocol is H.248. This termination is not involved unless special remarks are provided.)

When adding a user that supports terminal layering, you cannot and do not have to specify the parameter **terminalid** because the system automatically allocates a terminal ID. When adding a user that does not support terminal layering, you must specify the parameter **terminalid**.

You can run the **display tid-format(h248)** command or **display tid-format(mgcp)** command to query the TID formats of various types of users on the current MG interface. In the query result, **template-index** indicates the index of the TID template that is bound to the type of users. Then, run the **display tid-template** command to check whether the TID template supports the layering configuration. Hence, you can check whether the user supports terminal layering.

Precaution

- There are 19 default TID templates (templates 0-18) in the system. The default TID templates can be referenced, but cannot be added, modified, or deleted.
- The configuration of terminal layering on the MG must be the same as that on the MGC.
- If certain type of terminals exists on an interface and the interface is not disabled, the terminal prefix of this type of terminals cannot be modified.
- If a certain type of terminals exists on an interface, the TID format (including the terminal prefix and the index of the TID template) of this type of terminals cannot be changed.
- The terminal prefix must comply with the following syntax rules: The prefix must not exceed 64 characters. Only letters, digits, "_", and "/" are the characters allowed for input. The first character must be a letter, and the last character must not be a digit.
- The length of the TID, which is generated by combining the TID template and the terminal prefix that you configured, must not exceed 64 characters.

Procedure

- If the system protocol is H.248, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface h248** command to enter the H.248 mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).

NOTICE

The terminal prefix of PSTN, BRA and PRA must be all identical or unique.

  - In the H.248 mode, run the **tid-format rtp** command to configure the TID template and the terminal prefix of the RTP ephemeral termination.
  - In the H.248 mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
  - In the H.248 mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
  - In the H.248 mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.

  e. Run the **display tid-format(h248)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.
- If the system protocol is MGCP, to configure the TID format on the current MG interface, do as follows:
  a. Run the **display tid-template** command to query the information about the default TID template of the system.
  b. If the default TID template cannot meet the service requirements, run the **tid-template add** command to add a TID template that meets the service requirements. If the default TID template meets the service requirements, proceed to c.
  c. Run the **interface mgcp** command to enter the MGCP mode.
  d. Configure the TID templates and the terminal prefix of various types of users on the current MG (VAG).
     - In the MGCP mode, run the **tid-format pstn** command to configure the TID template and the terminal prefix of the PSTN user.
     - In the MGCP mode, run the **tid-format bra** command to configure the TID template and the terminal prefix of the ISDN BRA user.
     - In the MGCP mode, run the **tid-format pra** command to configure the TID template and the terminal prefix of the ISDN PRA user.

e. Run the **display tid-format(mgcp)** command to check whether the TID templates and the terminal prefixes of various types of users on the current MG interface are the same as those in the data plan.

----End

Example

Assume that in the H.248 mode, the terminal prefix of the PSTN user on MG interface 1 is aln/, and the layering TID template 3 is used. To add a PSTN user on port 0/2/0 and check whether the system automatically allocates a TID generated according to the template, do as follows:

```
huawei(config)#display tid-template 3//Query the information about TID template 3
  --------------------------------------------------
  Index     : 3
  Format    : %u/%u/%u
  Para-list  : F+1,S+1,P+1  //The parameter list of the TID template includes keyword
"F", "S",
                //and "P", which indicates that this template supports terminal layering.
  Name      : Aln_Not_Fixed_1
  --------------------------------------------------
huawei(config)#interface h248 1
huawei(config-if-h248-1)#tid-format pstn prefix aln template 3
huawei(config-if-h248-1)#quit
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser add 0/2/0 1
huawei(config-esl-user)#display mgpstnuser 0/2/0
{ <cr>|endframeid/slotid/portid<S><1,15> }:

  Command:
         display mgpstnuser 0/15/0
  -------------------------------------------------------------------------
  F  /S /P    MGID    TelNo          Priority PotsLineType TerminalID
  -------------------------------------------------------------------------
  0  /2 /0    1       -             Cat3     DEL        aln/1/3/1
           //The system allocates the terminal ID according to the TID format.
  -------------------------------------------------------------------------
```

1.23.1.6 Enabling an MG Interface

Enabling an MG interface is to reset an MG interface to make the MG interface register with the MGC (or to make the modified attributes of the MG interface take effect) after the configuration of the MG interface is complete, so that the MG interface can work in the normal state.

Precaution

⚠ CAUTION

For the MG interface that has been in service, this operation interrupts the ongoing services carried on the MG interface. Hence, exercise caution when performing this operation.

Procedure

- Enable the MG interface that adopts the H.248 protocol.
  a. Run the **interface h248** command to enter the H.248 mode.
  b. Run the **reset coldstart** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-h248 all** command to check whether the MG interface is in the normal state.
- Enable the MG interface that adopts the MGCP protocol.
  a. Run the **interface mgcp** command to enter the MGCP mode.
  b. Run the **reset** command to enable the MG interface.
  c. Run the **quit** command to return to the global config mode, and then run the **display if-mgcp all** command to check whether the MG interface is in the normal state.

----End

Example

To enable H.248-based MG interface 0, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-h248-0)#quit
huawei(config)#display if-h248 all
--------------------------------------------------------------------------------
MGID    Trans    State         MGPort MGIP         MGCPort MGCIP/DomainName
--------------------------------------------------------------------------------
0       UDP      Normal        2944   10.10.10.11    2944   10.10.20.11
--------------------------------------------------------------------------------
```

To enable MGCP-based MG interface 0, do as follows:

```
huawei(config)#interface mgcp 0
huawei(config-if-mgcp-0)#reset
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-mgcp-0)#quit
huawei(config)#display if-mgcp all
  --------------------------------------------------------------------------
  MGID    State         MGPort MGIP          MGCPort MGCIP/DomainName
  --------------------------------------------------------------------------
  0       Normal        2727   10.10.10.11    2727    10.10.20.11
  1       Wait ack      2527   10.10.10.12    2727    10.10.20.12
  --------------------------------------------------------------------------
```

1.23.2 Configuring the IUA Link

This topic describes how to configure the IUA link for signaling transmission between the Access node and MGC in the VoIP ISDN   service.

Context

- Simple Control Transmission Protocol (SCTP) is a connection-oriented protocol. Its most fundamental function is to provide reliable transmission for interaction messages between the Access node and MGC. The SCTP protocol implements services based on the association between two SCTP endpoints. SCTP can be regarded as a transmission layer. Its upper layer is called SCTP subscriber, and its lower layer is the IP network.
- The IUA link is the carrier of the interaction signaling between the Access node and MGC.

1.23.2.1 Adding an IUA Link Set

This topic describes how to add an IUA link set. When configuring the VoIP ISDN service, you need to configure the IUA link to carry the Q.931 call signaling. Before adding an IUA link, you must add a corresponding link set. Otherwise, the link cannot be added.

Context

- The system supports the configuration of a maximum of IUA link sets.
- After a link set is configured successfully, it is in the deactivated state by default.

Procedure

Step 1 In global config mode, run the command **sigtran** to enter Sigtran mode.

Step 2 Run the **iua-linkset add** command to add an IUA link set.

Step 3 You can run the **display iua-linkset attribute** command to check whether the configured IUA link set information is the same as the data plan.

----End

Example

Assume that the link set ID is 0, working mode of the link set is active/standby mode, pending duration is 20s, prefix of the IID is b/, IID generation mode is using the binary value that is automatically generated in ffsspp mode, namely, parameter 2. To add such an IUA link set, do as follows:

```
huawei(config)#sigtran
huawei(config-sigtran)#iua-linkset add 0 trafficmode override pendingtime 20 iid-map
2
braprefix b/
huawei(config-sigtran)#display iua-linkset attribute
{ <cr>|linksetno<L> }:

  Command:
         display iua-linkset attribute
----------------------------------------------------------
  LinksetNo           :0
  PendingTime          :20
  TrafficMode          :override
  C/S-Mode            :server
  IID-Type            :integer
  IID-Map             :2
  BRA IID-Prefix       :b/
```

```
 BRA IID-Suffix       :-
----------------------------------------------------------
```

1.23.2.2 Adding an IUA Link

This topic describes how to add an IUA link. After the link set is added, you can add an IUA link to carry the Q.931 call signaling for the ISDN user.

Prerequisites

The IUA link set must be added.

Context

- Make sure that a minimum of one item in the local port ID, local IP address, remote port ID, and remote IP address of a link is different from the corresponding item of other links.
- Only two links can be configured in the same link set. In addition, the local IP addresses of the two links must be the same.

Procedure

Step 1 (This step is not required if the command line interface is already in the Sigtran mode.) In global config mode, run the **sigtran** command to enter the Sigtran mode.

Step 2 Run the **iua-link add** command to add an IUA link.

Step 3 You can run the **display iua-link attribute** command to check whether the configured IUA link information is the same as the data plan.

----End

Example

Assume that the link ID is 0, link set ID is 0, local port ID is 1402, local IP address is 10.10.10.10, remote port ID is 1404, and remote IP address 1 is 10.10.10.20. To add such a link, do as follows:

```
huawei(config)#sigtran
huawei(config-sigtran)#iua-link add 0 0 1402 10.10.10.10 1404 10.10.10.20
huawei(config-sigtran)#display iua-link attribute
{ <cr>|linkno<L> }:

  Command:
         display iua-link attribute
  ----------------------------------------------------------------------
  LinkNo              :  0
  LinksetNo           :  0
  Local port          :  1402
  Local IP address     :  10.10.10.10
  Remote port         :  1404
  Remote IP address    :  10.10.10.20
  Remote IP address 2   :  -
  Priority            :  0
  ----------------------------------------------------------------------
```

1.23.3 Configuring the VoIP ISDN PRA User

This topic describes how to configure the VoIP ISDN PRA user. After the MG interface is configured, you can add the VoIP ISDN PRA user on this interface to implement the VoIP ISDN PRA service.

1.23.3.1 Configuring the Attributes of the E1 Port

This topic describes how to configure the attributes of the E1 port to ensure that the ISDN PRA port meets the actual application requirements.

Context

You can configure the impedance, line coding mode, and working mode of the E1 port.

Default Configuration

Table 1-48 lists the default values of the E1 port. When configuring the attributes of the E1 port, you need to modify the values according to the service requirements.

Table 1-48 Default values of the E1 port

Parameter	Default Setting
Port impedance	E1 mode: 75 ohm
Line coding mode	E1 mode: HDB3
CRC4	Enable
The mode for digital section access	Digital
Signaling type	CCS

Procedure

Step 1 In global config mode, run the **interface edt** command to enter the EDT mode.

Step 2 (Optional) Run the **e1port impedance** command to configure the impedance of an E1 port.

Step 3 (Optional; perform this step when you need to modify the line coding mode of the port) Run the **e1port line-code** command to configure the line coding mode of the E1 port.

NOTE

In E1 mode, the system supports two line coding modes, namely, HDB3 and AMI.

Step 4 (Optional) Run the **e1port crc4** command to configure the CRC4 function of an E1 port.

Step 5 (Optional) Run the **e1port attribute set** command to configure the digital section access mode of an E1 port.

Step 6 (Optional) Run the **e1port signal** command to configure the signaling type of an E1 port.

----End

Example

Assume that the E1 ports on ISDN PRA board work in HDB3 line encoding mode, the CRC4 function is enable, To configure such E1 ports, do as follows:

```
huawei(config)#interface edt 0/1
huawei(config-if-edt-0/1)#e1port line-code 1 HDB3
huawei(config-if-edt-0/1)#display e1port line-code 1
  ----------------------
  F/S/P   linecode
  ----------------------
  0/1/1   HDB3
  ----------------------
huawei(config-if-edt-0/1)#e1port crc4 1 enable
huawei(config-if-edt-0/1)#display e1port attribute 1
  --------------------------------------------------------------
   F/S/P    Signaltype    CRC4    Impedance    Accessmode
  --------------------------------------------------------------
   0/1/1    CCS          Enable  75          Digital
  --------------------------------------------------------------
```

1.23.3.2 Configuring the ISDN PRA User Data

This topic describes how to configure the ISDN PRA user data on the H.248 interface (the data must be the same as the corresponding data on the MGC) so that the ISDN PRA user can access the network to use the ISDN PRA service.

Prerequisites

The ISDN PRA service board must be inserted into the planned slot correctly and the board must be in the normal state.

Default Configuration

Table 1-49 lists the default settings of the attributes of the ISDN PRA user. When configuring these attributes, you can modify the values according to the service requirements.

Table 1-49 Default settings of the attributes of the ISDN PRA user

Parameter	Default Settings
Priority of the ISDN PRA user	cat3: (common user)
Flag of reporting the UNI fault of the ISDN PRA user	Disable
Sub-channel active mask of the ISDN PRA user	255.255.255.255
Threshold for the number of auto recoveries from deterioration faults	20

Procedure

Step 1 In global config mode, run the **esl user** command to enter the ESL user mode.

Step 2 Run the **mgprauser add** command to add an ISDN PRA.

- When **iid-map** in the **iua-linkset add** command is configured to 1, interfaceid must be configured and be different from the interfaceid of other users in the same link set.
- The terminal ID of an ISDN PRA user must be unique.
- Each ISDN PRA user occupies 32 terminal IDs. You need to input only the first terminal ID when adding an ISDN PRA user. If the MG interface does not support the terminal layering function, plus (or minus) at least 32 to (or from) the terminal ID of the previous ISDN PRA user when adding an ISDN PRA user, and use the result as the first terminal ID of the current ISDN PRA user.
- If the MG interface supports the terminal layering function, the terminal ID cannot be configured when an ISDN PRA user is added on the MG interface. The system automatically allocates a terminal ID for the user.

Step 3 Run the **display mgprauser** command to check whether the ISDN PSTN user data is the same as the data plan.

Step 4 (Perform this step when you need to modify the attributes of an ISDN PRA user.) Run the **mgprauser attribute set** command to configure the attributes of the ISDN PRA user.

Step 5 (Perform this step only after you modify the attributes of the ISDN PRA user.) Run the **display mgprauser** command to query whether the configured attributes of the ISDN PRA user are the same as the data plan.

----End

Example

Assume that the MG interface ID is 0, link set ID is 0, IUA port ID is 0 (iid-map is 1), terminal ID is 223 (does not support the terminal layering function), user priority is cat2, UNI alarm report flag is enable, sub-channel active mask is 255.255.255.255, and the threshold for the number of auto recoveries from L1 is 30. To add such an ISDN PRA user connected to the 0/1/0 port, do as follows:

```
huawei(config)#esl-user
huawei(config-esl-user)#mgprauser add 0/1/0 0 0 interfaceid 0 terminalid 223
huawei(config-esl-user)#display mgprauser 0/1
  --------------------------------------------------------------
  F  /S /P /B   MGID    LinkSetNo   InterfaceID   TerminalID
  --------------------------------------------------------------
  0/1/0 /0   0       0         0           A223
  --------------------------------------------------------------

huawei(config-esl-user)#mgprauser attribute set 0/1/0 priority cat2 unireport en
able activemask 255.255.255.255 auto-resume-limit 30
huawei(config-esl-user)#display mgprauser attribute 0/1/0
{ <cr>|endframeid/slotid/portid<S><Length 1-15> }:

  Command:
         display mgprauser attribute 0/1/0
  --------------------------------------------------------------------------------
  F  /S /P                : 0/1/0
```

```
 UNIreport               : enable
 Prior                  : Cat2
 Mask of sub channel       : 255.255.255.255
 Auto reservice times/limit : 0/30
  ------------------------------------------------------------------------------
```

1.23.3.3 (Optional) Configuring the System Parameters

This topic describes how to configure the system parameters including the overseas version flag and message waiting indication (MWI) mode according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **system parameters** command to configure the system parameters.

Step 2 Run the **display system parameters** command to check whether the system parameters are the same as those in the data plan.

----End

Example

To configure the overseas version flag (system parameter 1) to Hong Kong (parameter value 1), do as follows:

```
huawei(config)#system parameters 1 1
huawei(config)#display system parameters 1
  -----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 1
Mean: Overseas version flag, 0:China, 1:HongKong, 2:Brazil, 3:Egypt,
4:Singapore, 5:Thailand, 6:France, 7:Britain MSFUK, 8:Britain ETSI, 9:Bulgaria,
10:Reserved, 11:Austria, 12:Hungary, 13:Poland, 14:Turkey, 18:Belarus,
20:Germany
  -----------------------------------------------------------------------------
```

1.23.3.4 (Optional) Configuring the Overseas Parameters

By default, the overseas parameters are configured according to the Chinese standards. In the actual service configuration, the attributes such as the upper and lower thresholds of the flash-hooking duration can be configured according to the local standards to ensure that the response of the user terminal complies with the local standards.

Procedure

Step 1 Run the **oversea parameters** command to configure the overseas parameters.

Step 2 Run the **display oversea parameters** command to check whether the overseas parameters are the same as those in the data plan.

----End

Example

To set the upper flash-hooking threshold (overseas feature parameter 0) to 800 ms (in compliance with the Hong Kong standard) and the lower flash-hooking threshold (overseas feature parameter 1) to 100 ms (in compliance with the Hong Kong standard), do as follows:

```
huawei(config)#oversea parameters 0 800
huawei(config)#oversea parameters 1 100
huawei(config)#display oversea parameters
{ <cr>|name<U><0,99> }: 1

  Command:
          display oversea parameters 1
    ----------------------------------------------------------------------------
  Parameter name index: 1    Parameter value: 100
  Mean: Hooking lower threshold(ms), reference: China:100, HongKong:100
  ----------------------------------------------------------------------------
```

1.24 Configuring the R2 Service

With the R2 access technology, the Access node provides access services on common twisted pair cables when interconnecting with the PBX using R2 signaling.

Prerequisites

- The operation mode of the EDTB board must be configured as service mode, which can be configured using the **board workmode** command.
- The working mode of the EDTB board must be configured as voice mode, which can be configured using the **runmode** command.
- The signaling type of ports on the EDTB board must be configured as channel associated mode, which can be configured using the **e1port signal** command.

For the H.248-based MoIP service:

- The MG interface has been configured. For details, see 1.23.1 Configuring an MG Interface.

For the SIP-based MoIP service:

- A SIP interface has been configured. For details, see 1.20.1 Configuring the SIP Interface.

Context

- R2 signaling is channel associated signaling (CAS), which is international standard signaling based on E1 digital network.
- The Access node connects the PBX and NGN network using R2 signaling, achieving transition from the PSTN network to the NGN network.

Procedure

Step 1 Run the **r2 profile** command to add an R2 profile. Define the R2 signaling with a specific feature as an R2 profile which can be used when an R2 user is added.

Step 2 (Optional) Run the **profile attribute** command to configure the signaling type of the R2 profile.

Step 3 (Optional) Configure adaptation data of the R2 profile.

The ITU-T Q.400-Q.490 standard has defined R2 signaling standard, but different countries and regions implement R2 signaling in different ways. You do not need to change parameter values if the parameter values defined in the signaling standard of a country are consistent with the default values defined by the Access node. Otherwise, you need to change the parameter values based on actual conditions.

- Run the **address-receive attribute** command to configure the receive attribute of R2 addresses.
- Run the **address-send attribute** command to configure the transmit attribute of R2 addresses.
- Run the **profile attribute** command to configure the signaling type of the R2 profile.
- Run the **line-signaling attribute** command to configure the R2 line signaling attribute.
- Run the **register-signaling attribute** command to configure the R2 register signaling attribute.

Step 4 (Optional) Run the **multi-r2-adapt add** command to add parameters of the state machine of register signaling and line signaling in adaptation profiles for multiple countries. To comply with the R2 standards of different countries, parameter configuration for the R2 state machine is added, so that mappings between logical commands and physical commands can be changed by changing configurations without adding logical commands.

Step 5 Run the **mgr2user add** command (for H.248 protocol) or **sipr2user add** command (for SIP protocol) to add an R2 user.

Step 6 (Optional) Run the **mgr2user attribute set** command (for H.248 protocol) or **sipr2user attribute set** command (for SIP protocol) to set the priority of R2 users. When congestion occurs, packets of the user with a high priority is forwarded first.

----End

Example

For example, configure R2 users at the 0/2/1 port when using SIP protocol. Parameters are shown in Table 1-50.

Table 1-50 Data plan for R2 user configuration

Configuration Item	Data
R2 profile	0
Signaling type of the R2 profile	10
Wait-answer-time	200s
Wait-protect-time	300 ms
SIP interface	0
Subrack ID/slot ID/port number	0/2/1
The terminal priority of an R2	cat2

Configuration Item	Data
user	

```
huawei(config)#r2 profile 0
  Are you sure to add r2 profile?(y/n)[n]:y
huawei(config-r2-0)#profile attribute name normal signaling-type 10
huawei(config-r2-0)#line-signaling attribute wait-answer-time 200 wait-protect-time
300
huawei(config-r2-0)#quit
huawei(config)#esl user
huawei(config-esl-user)#sipr2user add 0/2/1 0 0
huawei(config-esl-user)#sipr2user attribute set 0/2/1 priority cat2
```

1.25 Configuring the H.248/MGCP-based FoIP Service

This topic describes how to configure the H.248/MGCP-based FoIP service.

Prerequisites

The VoIP service must be configured. For details, see 1.21 Configuring the VoIP PSTN Service (H.248-based or MGCP-based).

The voice service port used for the FoIP service must be in the normal state, and the voice communication on the port must be normal.

Context

Fax over Internet Protocol (FoIP) provides fax services on IP networks or between IP networks and PSTN networks. The FoIP service can be classified from two aspects:

- In terms of coding mode, the fax mode can be transparent fax (G.711 coding) or T.38 fax (T.38 coding).
- In terms of the participation of the MGC, one is the softswitch controlled flow, and the other is the self-switch flow (controlled by the gateway itself).

Data preparation

Before the data configuration, it is recommended that you plan the working mode of the FoIP service on the entire NGN network to ensure that the configurations on the entire network are consistent.

Table 1-51 Fax flows

Item	Flow	Remarks
Coding negotiation mode	Negotiation	The gateway negotiates the coding mode with the MGC through signaling.
	Self-switch	The gateway determines the coding mode to be adopted.

Item	Flow	Remarks
Coding mode	Transparent transmission fax	The G.711 coding mode is adopted.
	T.38 flow	The T.38 coding mode is adopted.
Negotiation flow	V2 flow (auto-negotiation flow)	The V2 flow is adopted as the fax/modem flow.
	V3 flow	The V3 flow is adopted as the fax/modem flow.
	V5 flow	The V5 flow is adopted as the fax/modem flow.

NOTE

V2, V3, and V5 flows refer to the versions of the fax/modem flow, which is defined by Huawei. If self-switch is adopted as the coding negotiation mode, the negotiation flow does not need to be configured.

Default Configuration

Table 1-52 lists the default settings of the FoIP flow.

Table 1-52 Default settings of the FoIP flow

Item	Default Setting
Coding negotiation mode	Negotiation
Coding mode	Transparent transmission fax
Negotiation flow	V3 flow
Enable packet interval of fax and modem to use only 10 ms or not	Disable
RFC2198 startup mode	DisableRfc2198SmartStartup
Event transmit mode	ControlledByMGC

Procedure

Step 1 Configure public fax and modem parameters.

In the global config mode, run the **fax-modem parameters negomode** command to configure public fax and modem parameters. The purpose of this step is to configure the coding negotiation mode. Two options are available: negotiation and self-switch.

Step 2 Configure the fax coding mode and negotiation flow.

In the global config mode, run the **fax parameters** command to configure the fax transmission mode. There are three key parameters, which are described as follows:

- **transmode**: The value 0 indicates the transparent transmission mode with the G.711 coding; the value 1 indicates the T.38 mode, which is a coding mode dedicated to the fax service. The default value is 0.
- **flow**: Options are V2, V3, and V5. The default value is V3.
- **is-port+2**: This parameter should be consistent with the T.38 fax port configured on the peer MGC. When **transmode** is T.38 and **flow** is V2, this parameter must be configured.

NOTICE

In the high-speed fax mode, the fax mode cannot be configured as the auto-negotiation (V2 flow) T.38 mode or the self-switch transparent transmission mode. If the fax mode is the auto-negotiation (V2 flow) T.38 mode or the self-switch transparent transmission mode, modify the configuration according to step 1 and step 2.

Step 3 Query the common parameters of the fax and modem or the fax parameter configuration.

1. Run the **display fax-modem parameters** command to query the fax negotiation flow.
2. Run the **display fax parameters** command to query the fax coding mode.

----End

Example

To configure the negotiation mode of the FoIP service on the MA5600T/MA5603T/MA5608T to negotiation, enable 10 ms packetization, enable RFC2198 smart startup mode, configure the event transfer mode of fax to RFC2833, and the fax working mode of the MA5600T/MA5603T/MA5608T to thoroughly, and configure the fax flow to V2 flow, do as follows:

```
huawei(config)#fax-modem parameters negomode selfswitch packet-interval-10ms en
able rfc2198-start-mode enableRfc2198SmartStartup transevent rfc2833
huawei(config)#fax parameters flow v2 workmode thoroughly
huawei(config)#display fax-modem parameters
  -----------------------------------------------------------------------------
  Negomode              : Self switch
  Packet-interval-10ms   : Enable
  Rfc2198-start-mode    : Enable Rfc2198SmartStartup
  TransEvent            : RFC2833
  Vbd-codec             : G.711A
  Vbd-payload-type       : Static
  -----------------------------------------------------------------------------

huawei(config)#display  fax parameters
  -----------------------------------------------------------------------------
  FAX transfers mode                :Thoroughly
  T38 Fax Port                     :RTP port
  FAX flow                        :V2 Flow
  -----------------------------------------------------------------------------
```

1.26 Configuring the SIP-based FoIP Service

This topic describes how to configure the SIP-based FoIP service.

Prerequisites

- The SIP-based VoIP service must be configured. For details, see 1.18 Configuring the VoIP PSTN Service (SIP-based).
- The voice service port used for the FoIP service must be in the normal state, and the voice communication on the port must be normal.

Context

According to the fax coding mode, the FoIP service is classified into two modes:

- Transparent transmission fax: uses the G.711 coding
- T.38 fax: uses the T.38 coding

In the fax service application, according to whether the SIP signaling is involved in controlling the transmission, the FoIP service is classified into two modes:

- Negotiate mode, in which the SIP signaling is involved in controlling the transmission
- Self-switch mode, in which the SIP signaling is not involved in controlling the transmission

Data preparation

Before the data configuration, it is recommended that you plan the working mode of the FoIP service on the entire IMS network to ensure that the configurations on the entire network are consistent.

Table 1-53 Fax flows

Item	Flow	Remarks
Coding negotiation mode	Negotiate	The gateway negotiates the coding mode with the IMS through SIP signaling.
	Self-switch	The gateway determines the coding mode to be adopted.
Coding mode	Transparent transmission fax	The G.711 coding mode is adopted.
	T.38 flow	The T.38 coding mode is adopted.

Default Configuration

Table 1-54 lists the default settings of the FoIP flow.

Table 1-54 Default settings of the FoIP flow

Item	Default Setting

Item	Default Setting
Coding negotiation mode	negotiate
Transmission mode	1: thoroughly
Packetization interval	1: 10ms
Codec mode of VBD	G.711A
Payload type of VBD	static
Attribute type of VBD	V.152

Procedure

Step 1 Configure the common parameters of fax and modem.

1. In the global config mode, run the **interface sip** command to enter the SIP interface mode.
2. In the SIP interface mode, run the **fax-modem parameters** command to configure the coding negotiation mode.

   The purpose of this step is to configure the coding negotiation mode. Key parameter **negomode**: includes the self-switch mode and the negotiate mode. By default, the negotiate mode is adopted.

Step 2 Configure the fax coding mode.

In the SIP interface mode, run the **fax parameters** command to configure the fax transmission mode. There is only one key parameter, which is described as follows:

transmode: The value 0 indicates the transparent transmission mode with the G.711 coding; the value 1 indicates the T.38 mode, which is a coding mode dedicated to the fax service. By default, the value 0 is adopted.

Step 3 Query the common parameters of the fax and modem or the fax parameter configuration.

1. Run the **display fax-modem parameters** command to query the fax negotiation flow.
2. Run the **display fax parameters** command to query the fax coding mode.

----End

Example

To configure the negotiation mode of the FoIP service on the SIP interface 0 to negotiate, use 20 ms packetization, configure the codec mode of VBD to G.711A, payload type of VBD to static, configure the attribute type of VBD to ietf, and the fax transmission mode to thoroughly, do as follows:

```
huawei(config-if-sip-0)#fax-modem parameters negomode negotiate rtp-interval 2
vbd-codec G.711
A vbd-pt-type static vbd-attribute-type ietf
huawei(config-if-sip-0)#fax parameters transmode 0
huawei(config-if-sip-0)#display fax-modem parameters
 ------------------------------------------------------------------------------
 MGID                  :0
 Nego-mode              :negotiate
 Rtp-interval            :20ms
 Vbd-codec              :G.711A
 Vbd-pt-type             :static
 Vbd-attribute-type        :ietf

 -------------------------------------------------------------------------------

huawei(config-if-sip-0)#display fax parameters
 ---------------------------------------
 MGID     Transmode
 ---------------------------------------
 0       Thoroughly
 ---------------------------------------
```

1.27 Configuring the MoIP Service

This topic describes how to configure the H.248/MGCP/SIP-based MoIP service for transmitting the traditional narrowband modem data service over the IP network.

Prerequisites

For the H.248/MGCP-based MoIP service:

- The MG interface must be configured. For details, see 1.23.1 Configuring an MG Interface.
- The VoIP users must be configured. For details, see 1.21.2 Configuring the VoIP PSTN User.

For the SIP-based MoIP service:

- The SIP interface must be configured. For details, see 1.20.1 Configuring the SIP Interface.
- The VoIP users must be configured. For details, see 1.21.2 Configuring the VoIP PSTN User.

Context

The MoIP service can be transmitted in two modes:

- One is the transparent transmission mode, also called the voice-band data (VBD) transparent transmission. In this mode, the MG adopts the G.711 coding to encode and decode modem signals, and processes modem signals as common RTP data. In other words, the MG does not process modem signals, and the modem modulation signals are transparently transmitted over the IP network through the VoIP channel.
- The other is the redundancy mode, also called the relay mode.

NOTICE

Currently, the MA5600T/MA5603T/MA5608T supports the modem service only in the transparent transmission mode.

The modem event report mode is classified into the delay mode, direct mode, and high-speed signal immediate mode.

- In the delay mode, the MA5600T/MA5603T/MA5608T does not report the modem event immediately after receiving an event. Instead, it waits for a period of time until the event times out and no V21flag event is reported. In this manner, when the high-speed fax machines fail in the high-speed transmission (the modem mode on the host) negotiation, the low-speed transmission mode (the fax mode on the host) can still be used for transmitting data.
- In the direct mode, the MA5600T/MA5603T/MA5608T reports the modem event to the MGC immediately after receiving the event from the drive. To enable the MGC to quickly respond to a modem event, configure the modem event report mode to the direct mode.
- In the high-speed signal immediate mode, the MA5600T/MA5603T/MA5608T reports low-speed modem signals after a delay of 5.5s and reports high-speed modem signals without delay.

The configuration of the MoIP service is mainly the configuration of the modem event report mode and the transmission mode. The default settings are direct mode and transparent transmission mode. If configuration is required, you only need to configure the event report mode. This is because currently the MA5600T/MA5603T/MA5608T supports the modem service only in the transparent transmission mode. Hence, you do not need to configure the transmission mode.

Procedure

- Configure the H.248/MGCP-based modem event report mode.

  In the global config mode, run the **modem parameters eventmode** command to configure the modem event report mode. By default, the direct mode is used.
- Configure the SIP-based modem event report mode.
  a. Run the **interface sip** command to enter the SIP interface mode.
  b. (Optional) Run the **modem parameters transmode** command to configure the modem event report mode.

----End

Example

To enable the MA5600T/MA5603T/MA5608T to communicate with the MGC through H.248, and configure the transmission mode of the modem to transparent transmission and the modem event report mode to delay mode, do as follows:

```
huawei(config)#modem parameters eventmode 0
huawei(config)#save
```

1.28 Adding a POTS IP SPC

A semi-permanent connection (SPC) exclusively occupies a voice channel to meet the communication requirements and to ensure the communication quality for particular and vital access subscribers. To configure an IP SPC, configure the data to set up a direct IP connection between the two ends of the VoIP service. In this manner, the voice media data can be directly transmitted to the peer end.

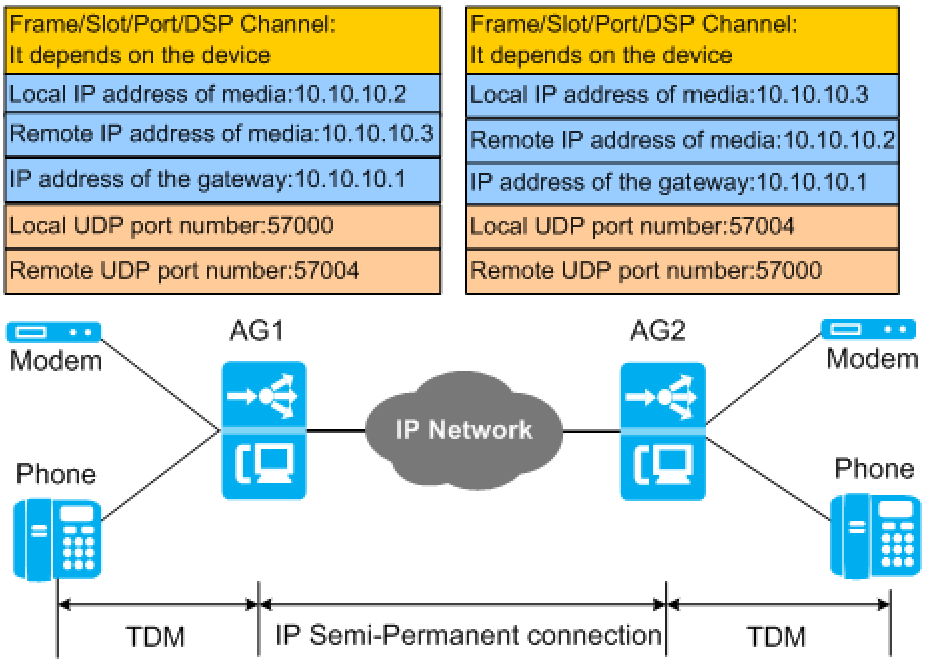
Prerequisites

- The electrical switch is already switched to the VoIP daughter board.
- The IP address of the VLAN interface is already configured.
- The remote VLAN interface can be routed and reached from the local VLAN interface.

Application Scenario

Figure 1-175 shows a sample network of the IP SPC service.

Figure 1-175 Network diagram of the IP SPC service


Frame/Slot/Port/DSP Channel:
It depends on the device
Local IP address of media:10.10.10.2
Remote IP address of media:10.10.10.3
IP address of the gateway:10.10.10.1
Local UDP port number:57000
Remote UDP port number:57004
Frame/Slot/Port/DSP Channel:
It depends on the device
Local IP address of media:10.10.10.3
Remote IP address of media:10.10.10.2
IP address of the gateway:10.10.10.1
Local UDP port number:57004
Remote UDP port number:57000
Modem
AG1
AG2
Modem
Phone
IP Network
Phone
TDM
IP Semi-Permanent connection
TDM

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Run the **ip address** command to configure the VoIP address pools.

Step 3 Run the **quit** command to quit the VoIP mode.

Step 4 Run the **spc** command to enter the SPC mode.

Step 5 Run the **ipspc add** command to add an SPC.

----End

Example

To add a POTS IP SPC, do as follows:

NOTE

The channel ID for a PSTN subscriber must be 0.

- Configuration procedure on AG1

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.2 10.10.10.1
huawei(config-voip)#quit
huawei(config)#spc
huawei(config-spc)#ipspc add 0/2/0/0 local-ip 10.10.10.2 local-port 57000 remote-ip
 10.10.10.3 remote-port 57004
```

- Configuration procedure on AG2

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.3 10.10.10.1
huawei(config-voip)#quit
huawei(config)#spc
huawei(config-spc)#ipspc add 0/2/0/0 local-ip 10.10.10.3 local-port 57004 remote-ip
 10.10.10.2 remote-port 57000
```

1.29 Adding a POTS IP SPC Hotline

A semi-permanent connection (SPC) hotline exclusively occupies a voice channel to meet the communication requirements and to ensure the communication quality for particular and vital access subscribers. To configure an IP SPC hotline, configure the data to set up a direct IP connection between the two ends of the VoIP service. In this manner, the voice media data can be directly transmitted to the peer end.

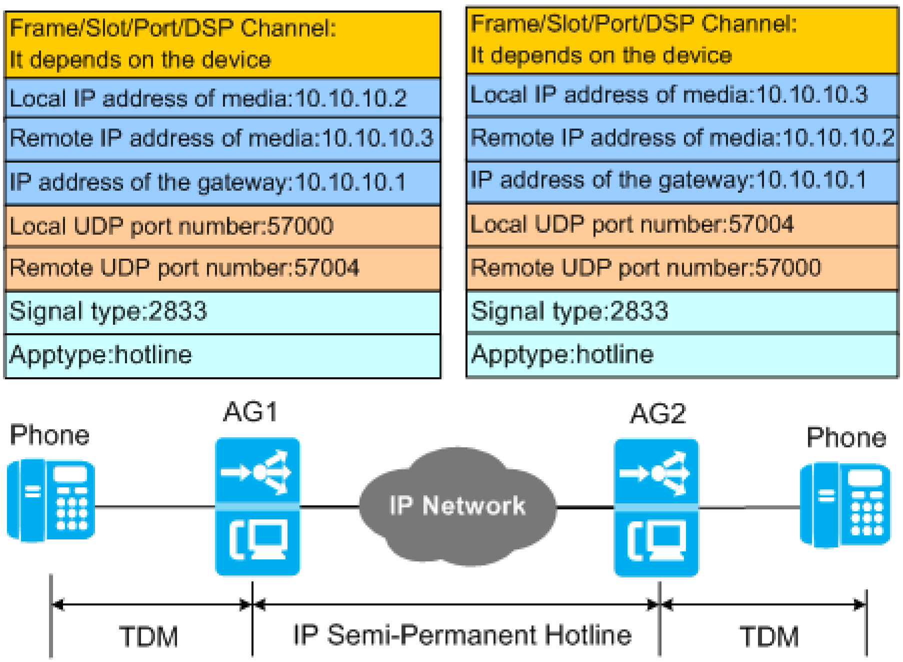
Prerequisites

- The electrical switch is already switched to the VoIP daughter board.
- The IP address of the VLAN interface is already configured.
- The remote VLAN interface can be routed and reached from the local VLAN interface.

Application Scenario

Figure 1-176 shows a sample network of the IP SPC hotline service.

Figure 1-176 Network diagram of the IP SPC hotline service



AG1	AG2
Frame/Slot/Port/DSP Channel: It depends on the device	Frame/Slot/Port/DSP Channel: It depends on the device
Local IP address of media:10.10.10.2	Local IP address of media:10.10.10.3
Remote IP address of media:10.10.10.3	Remote IP address of media:10.10.10.2
IP address of the gateway:10.10.10.1	IP address of the gateway:10.10.10.1
Local UDP port number:57000	Local UDP port number:57004
Remote UDP port number:57004	Remote UDP port number:57000
Signal type:2833	Signal type:2833
Apptype:hotline	Apptype:hotline

Procedure

Step 1 Run the **voip** command to enter the VoIP mode.

Step 2 Run the **ip address** command to configure the VoIP address pools.

Step 3 Run the **quit** command to quit the VoIP mode.

Step 4 Run the **spc** command to enter the SPC mode.

Step 5 Run the **ipspc add** command to add an SPC hotline.

----End

Example

To add a POTS IP SPC hotline, do as follows:

NOTE

The channel ID for a PSTN subscriber must be 0.

- Configuration procedure on AG_1

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.2 10.10.10.1
huawei(config-voip)#quit
huawei(config)#spc
huawei(config-spc)#ipspc add 0/2/0/0 local-ip 10.10.10.2 local-port 57000 remote-ip
 10.10.10.3 remote-port 57004 apptype hotline signal-type 2833
```

- Configuration procedure on AG_2

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.3 10.10.10.1
huawei(config-voip)#quit
huawei(config)#spc
huawei(config-spc)#ipspc add 0/2/0/0 local-ip 10.10.10.3 local-port 57004 remote-ip
 10.10.10.2 remote-port 57000 apptype hotline signal-type 2833
```

1.30 Configuring the IP Z Interface Extension Service

The following network typically applies to the scenario where Z interface extension private line service needs to be carried over the IP network for the headquarters (HQ) and branch offices of an enterprise after the PSTN network reconstruction. In the following configuration example, the FXO and FXS boards are added for the Z interface extension local MSAN and remote MSAN respectively, board attributes are configured, and IP semi-permanent connections (SPCs) of the IP Z interface extension type are created between the two boards, so that users connected to the FXS board are connected to the corresponding ports on the FXO board through the SPCs.

Prerequisites

IP Z interface extension is a technology proprietarily owned by Huawei. Therefore, the MSANs on the FXO side and the FXS side must be Huawei MSANs.

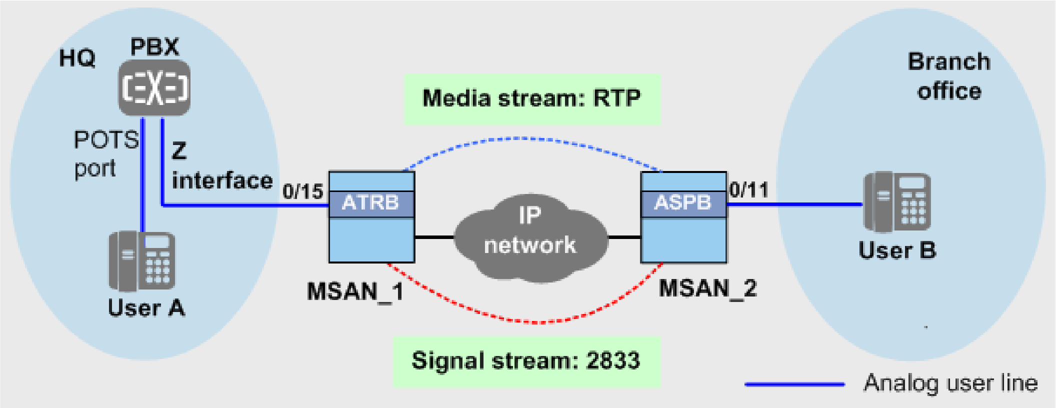
Application Scenario

NOTE

In this example, the MSANs at both ends of the IP network are Huawei MA5600T. The FXO board on MSAN_1 is the ATRB board, and the FXS board on MSAN_2 is the ASPB board. See more about the hardware support for the IP Z interface extension service , please refer to Introduction to IP Z Interface Extension .

Figure 1-177 shows a sample network of the IP Z interface extension service. The ATRB board of MSAN_1 connects to user A through the local exchange (the PBX) by using the Z interface. User B connects to the ASPB board of MSAN_2. The two MSANs are connected through the IP network. SPCs of the IP Z interface extension type are created between corresponding ports of the ASPB and ATRB boards to implement the IP Z interface extension service.

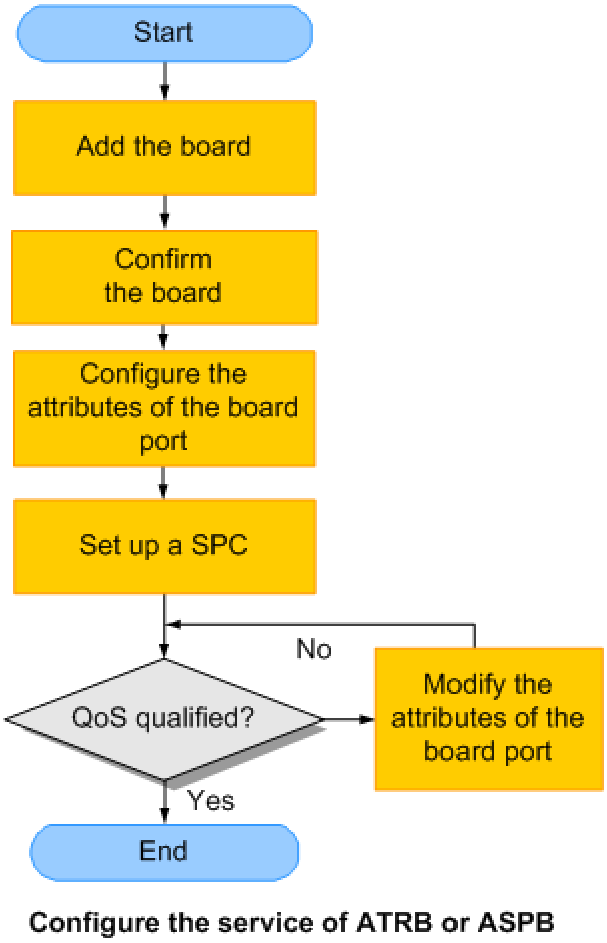
Figure 1-177 Network diagram of the IP Z interface extension service



Configuration Flow

Figure 1-178 shows the service configuration flows on the ATRB and ASPB boards.

Figure 1-178 Service configuration flows on the ATRB and ASPB boards


Start
Add the board
Confirm the board
Configure the attributes of the board port
Set up a SPC
No
QoS qualified?
Modify the attributes of the board port
Yes
End
Configure the service of ATRB or ASPB

Procedure

Step 1 Run the **board add** command to add an ATRB board on MSAN_1 and an ASPB board on MSAN_2.

Step 2 Run the **board confirm** command to confirm the ATRB board on MSAN_1 and the ASPB board on MSAN_2.

Step 3 on MSAN_1 and MSAN_2, run specially the **voip** command to enter the VOIP mode.

Step 4 on MSAN_1 and MSAN_2, run specially the **ip address** command to configure the IP address of the voice service.

Step 5 on MSAN_1 and MSAN_2, run specially the **quit** command to quit the VOIP mode.

Step 6 on MSAN_1, run the **fxoport** command to enter the FXO mode.

Step 7 on MSAN_1, run the **fxoport attribute set** command to configure port attributes for the ATRB board.

Step 8 on MSAN_1, run the**quit** command to quit the FXO mode.

Step 9 On MSAN_2, run the **pstnport** command to enter the PSTN mode.

Step 10 On MSAN_2, run the **pstnport_electric set** command to configure attributes of PSTN ports on the ASPB board.

Step 11 On MSAN_2, run the**quit** command to quit the PSTN mode.

Step 12 Run the **spc** command on MSAN_1 and MSAN_2 to enter the SPC configuration mode.

Step 13 Run the **ipspc add** command to configure SPCs of the IP Z interface extension type on MSAN_1 and MSAN_2.

----End

Example

Table 1-55 provides an example of related parameter configuration for the IP Z interface extension service.

Table 1-55 Data plan for the IP Z interface extension service

Configuration Item	Attribute	Data
ATRB board	Subrack ID/slot ID	0/15
ASPB board	Subrack ID/slot ID	0/11
VOIP address pools	Local IP address of media	10.10.10.2
	Remote IP address of media	10.10.10.3
	IP address of the gateway	10.10.10.1
ATRB port	Subrack ID/slot ID/port ID	0/15/0

Configuration Item	Attribute	Data
	Shortest duration for ringing current detection	2
	Threshold for ringing current message detection	4
ASPB port	Subrack ID/slot ID/port ID	0/11/0
	Rx gain on a port	0
SPC	Local IP address of the IP SPC	10.10.10.2
	Local UDP port number of the IP SPC	57000
	Remote IP address of the IP SPC	10.10.10.3
	Remote UDP port number of the IP SPC	57004
	Number of channels occupied by the IP SPC	1
	Signaling type of the IP SPC	2833 (Signaling is transmitted in the RFC2833 mode.)
	Initial ringing type of the IP SPC	Default value: 4 (This parameter is configured only on the port of the FXS board.)
	Cadence ringing type of the IP SPC	Default value: 0 (This parameter is configured only on the port of the FXS board.)

- Configuration procedure on MSAN_1

```
huawei(config)#board add 0/15 ATR
huawei(config)#board confirm 0/15
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.2 10.10.10.1
huawei(config-voip)#quit
huawei(config)#fxoport
huawei(config-fxoport)#fxoport attribute set 0/15/0  ring-detect
min-ontime 2 max-offtime 4
huawei(config-fxoport)#quit
huawei(config)#spc
huawei(spc)#ipspc add 0/15/0/0 local-ip 10.10.10.2 local-port 57000
remote-ip 10.10.10.3 remote-port 57004 signal-type 2833
```

- Configuration procedure on MSAN_2

```

huawei(config)#board add 0/11 ASP
huawei(config)#board confirm 0/11
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.3 10.10.10.1
huawei(config-voip)#quit
huawei(config)#pstnport
huawei(config-pstnport)#pstnport electric set 0/11/0 recvgain 0
huawei(config-pstnport)#quit
huawei(config)#spc
huawei(spc)#ipspc add 0/11/0/0 local-ip 10.10.10.3 local-port 57004
remote-ip 10.10.10.2 remote-port 57000 signal-type 2833 initialring 4
cadencering 0
```

NOTE

- The precautions of configuring Tx/Rx gains are shown as follows.
- You are advised to use default values for Tx gains of PSTN ports and FXO ports on the FXS board on MSAN_2 and those of PSTN ports on the PBX on MSAN_1. Do not change Tx gains of these ports to avoid the scenario in which dial tones cannot be cut because of Tx gain modification.
- You are advised to set the Rx gain of the PSTN port on the FXS board on MSAN_2 to its maximum 0dB (default value is -7dB), set that of the FXO port on the FXO board to -3dB, and set that of the PSTN port on the PBX on MSAN_1 to -4dB .
- In actual applications, use parameter **recvgain** in the **fxoport attribute set** command to configure the Rx gain of the FXO port and parameter **recvgain** in the **pstnport electric set** command to configure the Rx gain of the PSTN port on the FXS board.
- For details about the process and precautions of configuring ringing current detection parameters on the FXO port, see **1.14.5 Ringing and CLIP Services for IP Z Interface Extension Feature**.
- After configuring the port attributes, run the **display fxoport attribute** command to query the configuring information of port attributes.
- After configuring the SPCs of the IP Z interface extension, run the **display ipspc** command to query the configuring information of IP SPCs.

After the configuration takes effect, user A and user B can communicate.

1.31 Configuring VAGs

The purpose of configuring virtual access gateways (VAGs) is to simulate multiple AGs by using one AG, increasing the usage rate and flexibility of the device.

1.31.1 Configuring the VAG Service (H.248/MGCP)

This topic describes how to configure the VAG service by creating two MG interfaces on the MA5600T/MA5603T/MA5608T and configuring PSTN users on the two MG interfaces. This topic is applicable to the scenario where multiple logical AGs are simulated on one physical AG.

Context

Configuring VAGs is literally to configure MG interfaces with different IDs on the same device, and to configure user ports homing to different MG interfaces. Pay attention to the following points:

- When the system uses the MGCP or H.248 protocol, up to eight MG interfaces with different IDs can be configured on the MA5600T/MA5603T/MA5608T, and each MG interface can be considered as a VAG.
- When configuring the parameters for interconnecting different MG interfaces with the MGC, make sure that the values of the following parameters of the MG interfaces are not the same. The values of at least one of the following parameters must be different on the MG interfaces.
  - Local IP address
  - Local port ID
  - Remote IP address
  - Remote port ID

Service Requirements

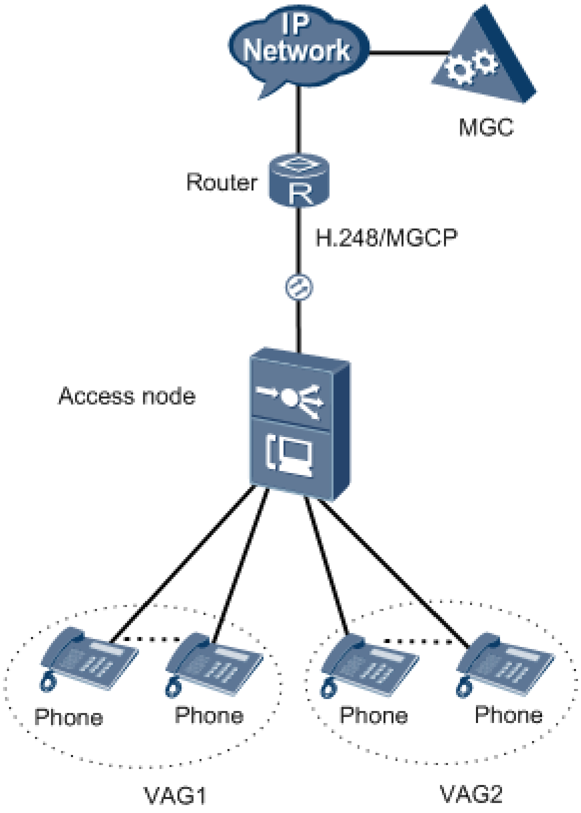
The service requirements are as follows:

- As shown in Figure 1-179, configure VAG1, and configure the users in slot 0/2 to belong to VAG1; configure VAG2, and configure the users in slot 0/3 to belong to VAG2.
- The MG communicates with the MGC through the H.248 protocol.
- Configure the data plan of VAG1 as listed in Table 1-56.
- The data plan of VAG2 is the same as that of VAG1 except for the following differences:
  - The media IP address and signaling IP address of VAG2 are 10.10.10.11.
  - The MGC assigns phone numbers 85110000-85110031 to phones 0-31 of VAG2.
- MG0 indicates VAG1 in the figure, and MG1 indicates VAG2 in the figure.

Networking

Figure 1-179 shows an example network of the VAG service.

Figure 1-179 Example network of the VAG service



Dataplan

Table 1-56 Data plan of VAG1

Item			Data
MG interface data (The data configuration must be the same as the data configuration on the MGC.)	Media and signaling parameters	Media and signaling upstream VLAN	Standard VLAN is recommended as the upstream VLAN of the voice service. Standard VLAN 20 is adopted.
		Media and signaling upstream port	In this office, all the services are transmitted upstream through the upstream interface board. Therefore, port 0/19/0 is used as the upstream port of the voice service.
		Media IP address and signaling IP address	These two IP addresses are both 10.10.10.10.
		Default gateway IP address	The IP address of the next hop from the MA5600T/MA5603T/MA5608T to the MGC is 10.10.10.1.
	Attributes	MG interface ID	0 and 1

Item			Data
	of the MG interface **NOTE** Parameters listed here are mandatory, which means that the MG interface fails to be enabled if these parameters are not configured.	Signaling port ID of the MG interface	2944
		IP address of the primary MGC to which the MG interface belongs	The IP address of the primary MGC is 10.10.20.20, and the port ID is 2944, the same as the port ID on the MA5600T/MA5603T/MA5608T.
		Port ID of the primary MGC to which the MG interface belongs	
		Coding mode of the MG interface	**text** (indicates the text coding mode)
		Transmission mode of the MG interface	UDP
	1.23.1.5 (Optional) Configuring the TID Format of an MG Interface		To differentiate users according to the TID, the office requires that the terminal prefix uses the community name **huawei** and the TID is automatically generated by the system according to the subrack ID/slot ID/port ID (F/S/P) of the user. Run the **display tid-template** command to query the default TID template. It is found that default TID template (template 6) can meet the requirements.
Voice user data (The data configuration must be the same as the data configuration on the MGC.)	Slot of the voice service board		User access is implemented by the ASPB board in slot 0/2 and 0/3.
	1.21.2.1 Configuring the PSTN User Data	Phone number	The emergency standalone is not supported, and therefore the phone numbers do not need to be configured when the users are added. The MGC assigns phone numbers 83110000-83110031 to phones 0-31.
		TID	The terminal layering is supported, and therefore the TIDs do not need to be allocated manually.

Procedure

- Configure VAG1.

  a. Add the upstream VLAN interface.

According to the data plan, configure standard VLAN 20 as the media and signaling upstream VLAN, add upstream port 0/19/0 to the VLAN, and configure the IP address of the Layer 3 interface to 10.10.10.10, which facilitates the configuration of the media and signaling IP address pools.

```
huawei(config)#vlan 20 standard
huawei(config)#port vlan 20 0/19 0
huawei(config)#interface vlanif 20
huawei(config-if-vlanif20)#ip address 10.10.10.10 24
huawei(config-if-vlanif20)#quit
```

b. Configure the media and signaling IP address pools.

Add the IP address of the VLAN Layer 3 interface configured in the previous step to the media and signaling IP address pools respectively. Therefore, the media and signaling IP addresses used for the services can be selected from the IP address pools. According to the data plan, IP address 10.10.10.10 is added to the media and signaling IP address pools, and the gateway IP address corresponding to the media IP address is 10.10.10.1.

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.10 10.10.10.1
huawei(config-voip)#ip address signaling 10.10.10.10
huawei(config-voip)#quit
```

c. Add an MG interface.

Add an MG interface to communicate with the MGC, which ensures that the MGC can control the call connection through the MG interface. According to the data plan, add MG interface 0, and configure the interface attributes.

```
huawei(config)#interface h248 0
  Are you sure to add MG interface?(y/n)[n]:y
huawei(config-if-h248-0)#if-h248 attribute mg-media-ip1 10.10.10.10 mgip
10.10.10.10 mgport 2944 primary-mgc-ip1 10.10.20.20
primary-mgc-port 2944 code text transfer udp
```

d. Configure the TID template of the PSTN users on MG interface 0.

Configure the TID generation mode. According to the data plan, the terminal prefix of PSTN users is **huawei**, and the TID template adopts layering template 6.

NOTICE

The MA5600T/MA5603T/MA5608T requires that the terminal prefixes of PSTN users, ISDN BRA users, and ISDN PRA users on the same H.248 interface be either the same or different. Note this when configuring the terminal prefix.

```
huawei(config-if-h248-0)#tid-format pstn prefix huawei template 6
```

e. Reset the MG interface.

Reset the MG interface to make the MG interface register with the MGC (and to make the modified attributes of the MG interface take effect) so that the MG interface can work in the normal state. The MG interface can be started in different ways (see Parameters of the **reset** command). For a newly configured MG interface, enable the MG interface through cold start.

```
huawei(config-if-h248-0)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-h248-0)#quit
```

f. Query the running status of the MG interface.

After the MG interface is interconnected with the MGC, the MG interface should be in the normal state, indicating that the MG interface can work in the normal state.

```
huawei(config)#display if-h248 all

-------------------------------------------------------------------------
----
  MGID    Trans State         MGPort MGIP          MGCPort MGCIP/DomainName

-------------------------------------------------------------------------
----
  0       UDP   Normal        2944   10.10.10.10    2944    10.10.20.20

-------------------------------------------------------------------------
----
```

g. Confirm the service board.

Confirm the ASPB service board that carries services so that the board can work in the normal state.

```
huawei(config)#board confirm 0/2
```

h. Configure the PSTN user data.

Add POTS users (phones 0-31) so that the users can go online.

```
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser batadd 0/2/0 0/2/31 0
huawei(config-esl-user)#quit
```

i. Save the data.

```
huawei(config)#save
```

- Configure VAG2.

a. Add the upstream VLAN interface.

According to the data plan, configure the secondary IP address of the Layer 3 interface to 10.10.10.11, which facilitates the configuration of the media and signaling IP address pools.

In step a, the VLAN is added and the IP address of the VLAN Layer 3 interface is configured. In this step, use the sub parameter to configure the secondary IP address of the VLAN. The secondary IP address is the same as the primary IP address in function. They are used for Layer 3 forwarding.

```
huawei(config)#interface vlanif 20
huawei(config-if-vlanif20)#ip address 10.10.10.11 24 sub
huawei(config-if-vlanif20)#quit
```

b. Configure the media and signaling IP address pools.

Add the IP address of the VLAN Layer 3 interface configured in the previous step to the media and signaling IP address pools respectively. Therefore, the media and signaling IP addresses used for the services can be selected from the IP address pools. According to the data plan, IP address 10.10.10.11 is added to the media and signaling IP address pools, and the gateway IP address corresponding to the media IP address is 10.10.10.1.

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.10.10.11 10.10.10.1
huawei(config-voip)#ip address signaling 10.10.10.11
huawei(config-voip)#quit
```

c. Add an MG interface.

Add an MG interface to communicate with the MGC, which ensures that the MGC can control the call connection through the MG interface. According to the data plan, add MG interface 1, and configure the interface attributes.

```
huawei(config)#interface h248 1
  Are you sure to add MG interface?(y/n)[n]:y
huawei(config-if-h248-1)#if-h248 attribute mg-media-ip1 10.10.10.11 mgip
10.10.10.11 mgport 2944 primary-mgc-ip1 10.10.20.20
primary-mgc-port 2944 code text transfer udp
```

d. Configure the TID template of the PSTN users on MG interface 1.

Configure the TID generation mode. According to the data plan, the terminal prefix of PSTN users is **huawei**, and the TID template adopts layering template 6.

NOTICE

The MA5600T/MA5603T/MA5608T requires that the terminal prefixes of PSTN users, ISDN BRA users, and ISDN PRA users on the same H.248 interface be either the same or different. Note this when configuring the terminal prefix.

```
huawei(config-if-h248-1)#tid-format pstn prefix huawei template 6
```

e. Reset the MG interface.

Reset the MG interface to make the MG interface register with the MGC (and to make the modified attributes of the MG interface take effect) so that the MG interface can work in the normal state. The MG interface can be started in different ways (see Parameters of the **reset** command). For a newly configured MG interface, enable the MG interface through cold start.

```
huawei(config-if-h248-1)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
huawei(config-if-h248-1)#quit
```

f. Query the running status of the MG interface.

After the MG interface is interconnected with the MGC, the MG interface should be in the normal state, indicating that the MG interface can work in the normal state.

```
huawei(config)#display if-h248 all

------------------------------------------------------------------------
----
  MGID    Trans State         MGPort MGIP          MGCPort MGCIP/DomainName

------------------------------------------------------------------------
----
   0       UDP   Normal         2944   10.10.10.10     2944    10.10.20.2
   1       UDP   Normal         2944   10.10.10.11     2944    10.10.20.20

------------------------------------------------------------------------
----
```

g. Confirm the service board.

Confirm the ASPB service board that carries services so that the board can work in the normal state.

```
huawei(config)#board confirm 0/3
```

h. Configure the PSTN user data.

Add POTS users (phones 0-31) so that the users can go online.

```
huawei(config)#esl user
huawei(config-esl-user)#mgpstnuser batadd 0/3/0 0/3/31 1
huawei(config-esl-user)#quit
```

i. Save the data.

```
huawei(config)#save
```

----End

Result

1. When MG0 communicates with the MGC in the normal state, phones 0-31 belonging to MG0 can communicate with each other.
2. When MG1 communicates with the MGC in the normal state, phones 0-31 belonging to MG1 can communicate with each other.
3. When MG0 and MG1 communicate with the MGC in the normal state, phones 0-31 belonging to MG0 and phones 0-31 belonging to MG1 can communicate with each other.

1.31.2 Configuring VAG Service (SIP)

This topic describes how to configure and verify the VAG service by creating two SIP interfaces on the MA5600T/MA5603T/MA5608T and configuring PSTN users on the two SIP interfaces. This topic is applicable to the scenario where multiple logical AGs are simulated on one physical SIP AG.

Context

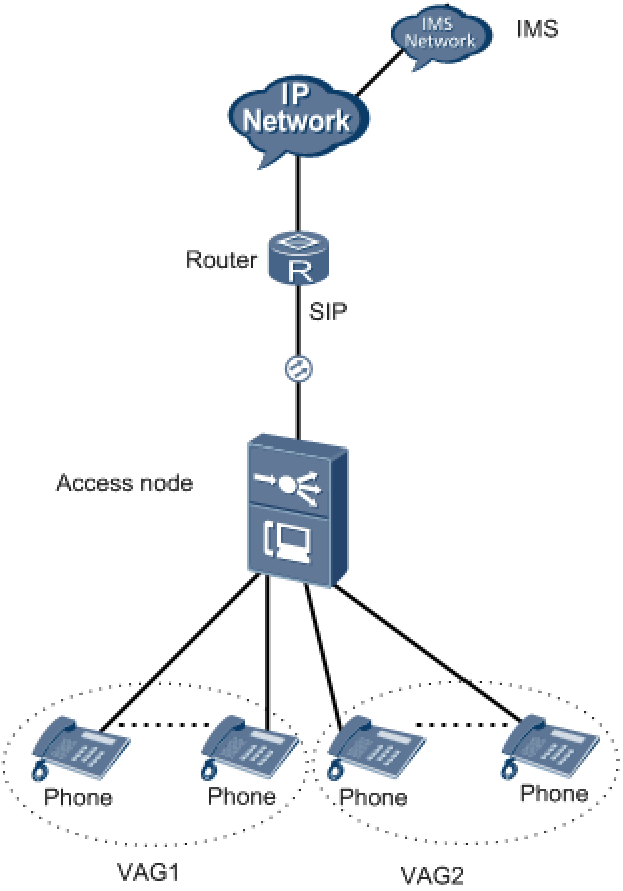
Configuring VAGs is literally to configure SIP interfaces with different IDs on the same device, and to configure user ports homing to different SIP interfaces. Pay attention to the following points:

- When the system uses the SIP protocol, up to eight SIP interfaces with different IDs can be configured on the MA5600T/MA5603T/MA5608T, and each SIP interface can be considered as a VAG.
- When configuring the parameters for interconnecting different SIP interfaces with the IMS, make sure that the values of the following parameters of the SIP interfaces are not completely the same. The values of at least one of the following parameters must be different on the SIP interfaces.
  - Local IP address
  - Local port ID
  - Remote IP address
  - Remote port ID

Service Requirements

Figure 1-180 shows an example network of the SIP-based VAG service.

Figure 1-180 Example network of the SIP-based VAG service



Data Plan

The service requirements are as follows:

- As shown in Figure 1-180, configure VAG1, and configure the users in slot 0/2 to belong to VAG1; configure VAG2, and configure the users in slot 0/3 to belong to VAG2.
- The MA5600T/MA5603T/MA5608T communicates with the IMS through the SIP protocol.
- Configure the data plan of VAG1 as listed in Table 1-57.
- The data plan of VAG2 is the same as that of VAG1 except for the following differences:
  - The media IP address and signaling IP address of VAG2 are 10.20.10.11.
  - The IMS assigns phone numbers 85000000-85000031 to phones 0-31 of VAG2.
- SIP interface 0 indicates VAG1 in the figure, and SIP interface 1 indicates VAG2 in the figure.

Table 1-57 Data plan of VAG1

Item			Remarks
SIP interface	Media and signaling	Media and signaling	Standard VLAN 30 is adopted.

Item			Remarks
data (The data configuration must be the same as the data configuration on the IMS.)	parameters	upstream VLAN	
		Media and signaling upstream port	In this office, all the services are transmitted upstream through the control board. Therefore, port 0/19/0 is used as the upstream port of the voice service.
		Media IP address and signaling IP address	These two IP addresses are both 10.20.10.10.
		Default gateway IP address	The IP address of the next hop from the MA5600T/MA5603T/MA5608T to the IMS core network device is 10.20.10.1.
	Attributes of the SIP interface **NOTE** Parameters listed here are mandatory, which means that the SIP interface fails to be enabled if these parameters are not configured.	SIP interface ID	0 and 1
		Signaling port ID of the SIP interface	5060
		IP address of the primary IMS core network device to which the SIP interface belongs	10.20.20.100
		Port ID of the primary IMS core network device to which the SIP interface belongs	5060
		Transmission mode of the SIP interface	UDP
		Homing domain name of the SIP interface	huawei
		Index of the profile used by the SIP interface	Default profile (profile 1)
Voice user data (The data configuration must be the same as the data configuration on the IMS.)	Slot of the voice service board		The ASPB service board in slot 0/2 and 0/3.
	Phone number		80000000-80000031

Procedure

- Configure VAG1.
  a. Add the upstream VLAN interface.

     According to the data plan, configure standard VLAN 30 as the media and signaling upstream VLAN, add upstream port 0/19/0 to the VLAN, and configure the IP address of the Layer 3 interface to 10.20.10.10, which facilitates the configuration of the media and signaling IP address pools.

```
huawei(config)#vlan 30 standard
huawei(config)#port vlan 30 0/19 0
huawei(config)#interface vlanif 30
huawei(config-if-vlanif30)#ip address 10.20.10.10 24
huawei(config-if-vlanif30)#quit
```

  b. Configure the media and signaling IP address pools.

     Add the IP address of the VLAN Layer 3 interface configured in the previous step to the media and signaling IP address pools respectively. Therefore, the media and signaling IP addresses used for the services can be selected from the IP address pools. According to the data plan, IP address 10.20.10.10 is added to the media and signaling IP address pools, and the gateway IP address corresponding to the media IP address is 10.20.10.1.

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.20.10.10 10.20.10.1
huawei(config-voip)#ip address signaling 10.20.10.10
huawei(config-voip)#quit
```

  c. Configure a static route to the IMS.

     Make sure that the route between the local device and the IMS is reachable. The static route is used as an example.

```
huawei(config)#ip route-static 10.20.0.0 255.255.0.0 10.20.10.1
```

  d. Add an SIP interface.

     According to the data plan, add SIP interface 0, and configure the interface attributes.

```
huawei(config)#interface sip 0
  Are you sure to add the SIP interface?(y/n)[n]:y
huawei(config-if-sip-0)#if-sip attribute basic media-ip 10.20.10.10 signal-ip
10.20.10.10 signal-port 5060 transfer udp primary-proxy-ip1 10.20.20.100
primary-proxy-port 5060
 home-domain huawei sipprofile-index 1
```

  e. Reset the SIP interface.

     Reset the SIP interface to make the SIP interface register with the IMS (and to make the modified attributes of the SIP interface take effect) so that the SIP interface can work in the normal state.

```
huawei(config-if-sip-0)#reset
  Are you sure to reset SIP interface?(y/n)[n]:y
```

  f. Confirm the service board.

Confirm the ASPB service board that carries services so that the board can work in the normal state.

```
huawei(config)#board confirm 0/2
```

g. Configure the PSTN user data.

Add POTS users (phones 0-31) to VAG1 so that the users can go online.

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser batadd 0/2/0 0/2/31 0 telno 80000000
huawei(config-esl-user)#quit
```

h. Save the data.

```
huawei(config)#save
```

- Configure VAG2.

a. Add the upstream VLAN interface.

According to the data plan, configure the IP address of the Layer 3 interface to 10.20.10.11, which facilitates the configuration of the media and signaling IP address pools.

When you need to connect the VLAN interface to multiple subnets and configure the secondary IP address, use **sub** parameter. The secondary IP address is the same as the primary IP address in function. They are used for Layer 3 forwarding.

In step a, the VLAN is added and the IP address of the VLAN Layer 3 interface is configured. In this step, use the sub parameter to configure the secondary IP address of the VLAN. The secondary IP address is the same as the primary IP address in function. They are used for Layer 3 forwarding.

```
huawei(config)#interface vlanif 30
huawei(config-if-vlanif30)#ip address 10.20.10.11 24 sub
huawei(config-if-vlanif30)#quit
```

b. Configure the media and signaling IP address pools.

Add the IP address of the VLAN Layer 3 interface configured in the previous step to the media and signaling IP address pools respectively. Therefore, the media and signaling IP addresses used for the services can be selected from the IP address pools. According to the data plan, IP address 10.20.10.11 is added to the media and signaling IP address pools, and the gateway IP address corresponding to the media IP address is 10.20.10.1.

```
huawei(config)#voip
huawei(config-voip)#ip address media 10.20.10.11 10.20.10.1
huawei(config-voip)#ip address signaling 10.20.10.11
huawei(config-voip)#quit
```

c. Configure a static route to the IMS.

Make sure that the route between the local device and the IMS is reachable. The static route is used as an example. If the static route has been configured, skip this step.

```
huawei(config)#ip route-static 10.20.0.0 255.255.0.0 10.20.10.1
```

d. Add an SIP interface.

According to the data plan, add SIP interface 1, and configure the interface attributes.

```
huawei(config)#interface sip 1
  Are you sure to add the SIP interface?(y/n)[n]:y
huawei(config-if-sip-1)#if-sip attribute basic media-ip 10.20.10.11 signal-ip
10.20.10.11 signal-port 5060 transfer udp primary-proxy-ip1 10.20.20.100
primary-proxy-port 5060
 home-domain huawei sipprofile-index 1
```

e. Reset the SIP interface.

Reset the SIP interface to make the SIP interface register with the IMS (and to make the modified attributes of the SIP interface take effect) so that the SIP interface can work in the normal state.

```
huawei(config-if-sip-1)#reset
    Are you sure to reset SIP interface?(y/n)[n]:y
```

f. Confirm the service board.

Confirm the ASPB service board that carries services so that the board can work in the normal state.

```
huawei(config)#board confirm 0/3
```

g. Configure the PSTN user data.

Add POTS users (phones 0-31) to VAG2 so that the users can go online.

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser batadd 0/3/0 0/3/31 1 telno 85000000
huawei(config-esl-user)#quit
```

h. Save the data.

```
huawei(config)#save
```

----End

Result

After the configuration:

- Users of VAG1 can communicate with each other.
- Users of VAG2 can communicate with each other.
- Users of VAG1 and VAG2 can communicate with each other.

1.32 Configuring the Security and Reliability of the Voice Service

The security configuration of the voice service includes the H.248-based, MGCP-based, or SIP-based device authentication configuration, and the reliability configuration of the voice service includes the dual-homing configuration and the emergency standalone configuration.

1.32.1 Configuring Device Authentication

Device authentication is a method to improve the security of the core network and prevent illegal devices from registering with the core network device.

1.32.1.1 Configuring Device Authentication (H.248-based)

This topic describes how to configure the H.248-based device authentication to prevent illegal MGs from registering with the MGC.

Prerequisite

- The MG interface must be configured successfully.
- The parameters, including the encryption type, the initial key and the DH authentication, and the MG ID, must be configured on the MGC. These parameters must be the same as the parameters configured on the MA5600T/MA5603T/MA5608T.

Precautions

If Huawei products such as the SoftX3000 is used as the MGC, the authentication MG ID must be a character string with more than eight bits.

Procedure

Step 1 In the global config mode, run the **interface h248** command to enter the MG interface mode.

Step 2 Run the **mg-software parameter 4** command to configure the registration mode.

Step 3 Run the **mg-software parameter 6 0** command to configure the device authentication function on the MG interface.

Step 4 Run the **auth** command to configure the authentication MG ID and the initial key.

Step 5 Run the **display auth** command to query the authentication parameters.

Step 6 Run the **reset coldstart** command to reset the MG interface.

Reset the MG interface to make the MG interface register with the MGC (and to make the modified attributes of the MG interface take effect) so that the MG interface can work in the normal state. The MG interface can be enabled in different ways (see Parameters of the **reset** command). For a newly configured MG interface, enable the MG interface through cold start.

----End

Example

Configure the authentication parameters for the MA5600T/MA5603T/MA5608T as listed in Table 1-58.

Table 1-58 Data plan for configuring the H.248-based authentication

Item	Data
MG ID	0
Whether the wildcard is used in the registration	Yes
Authentication MG ID	MA5600T/MA5603T/MA5608T. It must be the same as the authentication MG ID on the MGC. Otherwise, the MG cannot register with the MGC.

Item	Data
Initial key	0123456789ABCDEF. It must be the same as the initial key configured on the MGC.

The following is a configuration example based on the data plan:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-software parameter 4 0
huawei(config-if-h248-0)#display mg-software parameter 4
  -------------------------------------------------
  Interface Id:0         para index:4   value:0
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   4: Whether MG register to MGC with wildcard
     0: Yes
     1: No
huawei(config-if-h248-0)#mg-software parameter 6 0
huawei(config-if-h248-0)#display mg-software parameter 6
  -------------------------------------------------
  Interface Id:0         para index:6   value:0
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   6: Whether MG support authentication
     0: Yes
     1: No
huawei(config-if-h248-0)#auth auth_mgid MA5600T/MA5603T/MA5608T initial_key
0123456789ABCDEF
huawei(config-if-h248-0)#display auth
 [AUTH_PARA config]
  Initial Key   : 0123456789ABCDEF
  Auth MGid     : MA5600T/MA5603T/MA5608T
  Algorithm     : MD5
huawei(config-if-h248-0)#reset coldstart
  Are you sure to reset MG interface?(y/n)[n]:y
```

1.32.1.2 Configuring Device Authentication (MGCP-based)

This topic describes how to configure the MGCP-based authentication parameters for the MG interface on the MA5600T/MA5603T/MA5608T to implement device authentication and prevent illegal MGs from registering with the MGC.

Prerequisite

- The MG interface must be configured successfully.
- The parameters, including the encryption type, the initial key and the DH authentication, and the MG ID, must be configured on the MGC. These parameters must be the same as the parameters configured on the MA5600T/MA5603T/MA5608T.

Procedure

Step 1 In the global config mode, run the **interface mgcp** command to enter the MG interface mode.

Step 2 Run the **mg-software parameter 4** command to configure the registration mode.

Step 3 Run the **auth** command to configure the authentication MG ID and the initial key.

If Huawei products such as the SoftX3000 is used as the MGC, the authentication MG ID must be a character string with more than eight bits.

NOTE

When the MGCP protocol is used, the MG interface supports two authentication modes:

- Passive authentication mode: In this mode, the device registers with the MGC and is authenticated only after required by the MGC.
- Active authentication mode: In this mode, the device is authenticated when the device registers with the MGC.

In actual applications, you can select the authentication mode according to the requirements.

Step 4 Run the **display auth** command to query the authentication parameters.

Step 5 Run the **reset** command to reset the MG interface.

----End

Example

Configure the authentication parameters for the MA5600T/MA5603T/MA5608T as listed in Table 1-59.

Table 1-59 Data plan for configuring the MGCP-based device authentication

Item	Data
MG ID	0
Whether the wildcard is used in the registration	Yes
Authentication mode	Active authentication mode
Authentication MG ID	MA5600T/MA5603T/MA5608T. It must be the same as the authentication MG ID on the MGC. Otherwise, the MG cannot register with the MGC.
Initial key	0123456789ABCDEF. It must be the same as the initial key configured on the MGC.

The following is a configuration example based on the data plan:

```
huawei(config)#interface mgcp 0
huawei(config-if-mgcp-0)#mg-software parameter 4 0
huawei(config-if-mgcp-0)#display mg-software parameter 4
  -------------------------------------------------
  Interface Id:0         para index:4   value:0
  -------------------------------------------------
```

```
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   4: Whether MG register to MGC with wildcard
      0: Yes
      1: No
huawei(config-if-mgcp-0)#auth mode2 auth_mgid MA5600T/MA5603T/MA5608T initial_key
0123456789ABCDEF
huawei(config-if-mgcp-0)#display auth
  active request authentication mode config:
  Initial Key    : 0123456789ABCDEF
  Auth MGid      : MA5600T/MA5603T/MA5608T
  Algorithm      : MD5
huawei(config-if-mgcp-0)#reset
  Are you sure to reset MG interface?(y/n)[n]:y
```

1.32.1.3 Configuring Device Authentication Based on SIP

When the Session Initiation Protocol (SIP) is used, the voice service of the MA5600T/MA5603T/MA5608T supports the authentication for a SIP interface and single user in user name+password or user name+HA1 mode.

Prerequisite

- The SIP interface has been added. For details about how to add a SIP interface, see **1.20.1 Configuring the SIP Interface**.
- The authentication information has been configured on the IP multimedia subsystem (IMS) side.

Context

- The device authentication must be supported on the IMS side. Ensure that the authentication data on the device side is the same as that on the IMS side.
- The user authentication on the MA5600T/MA5603T/MA5608T running SIP involves SIP interface authentication and user authentication. In SIP interface authentication, proxy option detection messages are authenticated. In user authentication, user registration and call messages are authenticated.
- A SIP user can be authenticated based on a SIP interface or a single user. Run the **sip-auth parameter auth-mode** command to configure a user authentication mode.
  - If the user authentication mode is set to *interface*, only the user name and password configured based on a SIP interface can be used for user authentication when the user authentication is based on both a SIP interface and a single user.
  - If the user authentication mode is set to *single-user*, the user name and password configured based on a single user are preferentially used for user authentication when the user authentication is based on both a SIP interface and a single user. The default user authentication mode is *single-user*.

Procedure

- Perform the authentication based on a SIP interface.
  a. In the global config mode, run the **interface sip** command to enter the SIP interface mode.

  b. Run the **sip-auth-parameter** command to configure the authentication user name and password for the SIP interface.

     Security authentication information includes password authentication mode, user name, password, and user authentication mode.

     - Password authentication mode includes **password** and **ha1**. In **password** mode, the original user password is configured. In **ha1** mode, a password is generated after the original user password is encrypted by using the message digest 5 (MD5) algorithm.
     - User authentication mode includes **interface** and **single-user**. The **interface** mode indicates that authentication is performed based on interface. This means that all users under an interface share an authentication user name. The **single-user** mode indicates that each user has a unique identity.

  c. Run the **reset** command to reset the SIP interface.

- Perform the authentication based on a single user.
  a. In global config mode, run the **esl user** command to enter extend signaling link (ESL) user mode.
  b. According to the service type, run the **sippstnuser auth set** command or the **sipbrauser auth set** command or the **sipprauser auth set** command to configure the authentication user name, password for single user.
  c. Run the **display sippstnuser authinfo** command or the **display sipbrauser authinfo** command or the **display sipprauser authinfo** command to query the security authentication information.

----End

Example

Configure the security authentication information of SIP interface 0 on the MA5600T/MA5603T/MA5608T, where,

- User authentication mode is **interface**
- Password authentication mode is **password**
- User name is **huawei.com**
- Password is **123456789**

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#sip-auth-parameter auth-mode interface password-mode pas
sword
  User Name(<=64 characters, "-" indicates deletion):huawei.com
  User Password(<=64 characters, "-" indicates deletion):   //Enter password here.
  The configuration will take effect after resetting the interface
huawei(config-if-sip-0)#reset
  Are you sure to reset the SIP interface?(y/n)[n]:y
```

Configure the security authentication information of the PSTN user on port 0/2/1, where,

- Telephone number is 88810001
- Authentication password mode is **password**
- User name is **huawei**
- Password is **huawei123**

To configure the authentication data of such a PSTN user, do as follows:

```
huawei(config)#esl user
huawei(config-esl-user)#sippstnuser auth set 0/2/1 telno 88810001 password-mode
password
  User Name(<=64 characters, "-" indicates deletion):huawei
  User Password(<=64 characters, "-" indicates deletion):   //Enter password here.
```

1.32.2 Configuring Inner Standalone (H.248-based or SIP-based)

This topic describes how to configure the inner standalone. After the inner standalone is configured, the internal phones can call each other using the internal extension numbers even if the interface between the gateway and the softswitch is interrupted.

Context

- The MG interface supports the inner standalone function only when it uses the H.248 protocol.
- When the MG interface works in the inner standalone state, only the internal users of the MG interface can communicate in the normal state.
- To maintain the same user phone number in the standalone state as the one used in normal condition, configure the phone number on the MG to be the same as that on the MGC.

Prerequisite

- The voice service users are configured properly on the H.248/SIP interface and the users can call each other successfully.
- The user phone number of the H.248 interface/SIP interface is configured to be the same as that on the softswitch. (Command: **mgpstnuser modify** (H.248 interface)/**sippstnuser modify** (SIP interface)).

Procedure

- Configure inner standalone (based on H.248 protocol)
  a. Run the **mg-software parameter 11 1** command to configure the MG interface to support the inner standalone function.
  b. Run the **digitmap set inner** command to configure the internal digitmap.

  NOTE
  The configured digitmap should correspond to the user phone number.

  c. (Optional) Run the **standalone parameters** command to configure the inner standalone timers.
     - The inner standalone timers include the dial tone timer (default: 10s), the busy tone timer (default: 40s), and the ringing tone timer (default: 50s).
     - Generally, the default inner standalone timers can be used.
- Configure inner standalone (based on SIP protocol)
  a. Run the **mg-software parameter 2 1** command to configure the SIP interface to support the inner standalone function.

     By default, the SIP inner standalone function is disabled.
  b. (Optional) Run the **local-digitmap add** command to add a digitmap used in SIP inner standalone.

The system has a digitmap named **defaultnormal**, which can match any phone numbers. Therefore, a new digitmap does not need to be added unless otherwise required.

----End

Example

Assume that an incoming third-party call will interrupt the inner standalone call after the communication between MG 0 and the MGC recovers. To configure MG 0 to support the inner standalone, and set the internal digitmap to 1234xxxx, do as follows:

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#mg-software parameter 11 1
huawei(config-if-h248-0)#display mg-software parameter 11
  -------------------------------------------------
  Interface Id:0         para index:11  value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   11: Stand alone support
       0: None
       1: Inner
       2: Emergency
       3: Both
huawei(config-if-h248-0)#digitmap set inner 1234xxxx
huawei(config-if-h248-0)#display digitmap
  -----------------------------------------------------------------------------
  Inner digitmap                                   : 1234xxxx
  Emergency digitmap                               : -
  Urgent digitmap (for overload or bandwidth restrict)   : -
  Dualdial digitmap for card service                  : -
  -----------------------------------------------------------------------------
```

To configure SIP interface 0 to support the inner standalone (so that the internal phones whose numbers belong to the number segment of 07552856xxxx can call each other, if the communication between SIP interface 0 and the softswitch is interrupted), do as follows:

```
huawei(config)#display local-digitmap name defaultnormal

  Command:
        display local-digitmap name defaultnormal
  ------------------------------------------
  Name: defaultnormal
  Type: normal
  Body: x.S|Exx.S      //can match any phone numbers
  ------------------------------------------
huawei(config)#interface sip 0
huawei(config-if-sip-0)#mg-software parameter 2 1
huawei(config-if-sip-0)#display mg-software parameter 2
  -------------------------------------------------
  MGID:0         para index:2   value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
```

```
Parameter Index:  Interface software parameter name:
   2 : SAL Support
       0: No
       1: Yes
```

1.32.3 Configuring the Dual Homing (Multi-Homing)

This topic describes how to configure the H.248-based, MGCP-based, and SIP-based dual homing (multi-homing). Dual homing (multi-homing) is a measure that protects the softswitch/IMS against a crash and a disaster recovery mechanism against accidents (such as a fire in the telecommunications room, disconnection of the cable connected to the telecommunications room, and abnormal power supply).

Context

The working principle of multi-homing is as follows: an MG interface (for H.248 and MGCP) or a SIP interface is homed to multiple registration servers. If the primary server malfunctions, the interface is switched to a secondary server and then continues to provide services.

Dual homing is one type of multi-homing configured with only primary/secondary servers without a disaster-recovery server.

1.32.3.1 Configuring H.248-based Dual Homing (Multi-homing)

In the case of H.248, the MA5600T/MA5603T/MA5608T supports homing of a media gateway (MG) interface to the primary/secondary media gateway controllers (MGCs) and disaster-recovery MGC. When the primary MGC malfunctions, the MG interface will register with the secondary MGC and then the disaster-recovery MGC cyclically.

Context

Technically speaking, dual homing is a configuration in which an MG is homed to the primary MGC and secondary MGC. Multi-homing is a configuration in which an MG is homed to the primary MGC, secondary MGC, and disaster-recovery MGC. Multi-homing is an enhancement of dual homing. In a broad sense, dual homing is one type of multi-homing.

The MA5600T/MA5603T/MA5608T provides different application policies for the dual homing (multi-homing) by configuring MG homing parameters and MG software parameters.

Procedure

Step 1 Create an MG interface and configure MG interface parameters.

1. In the global config mode, run the **interface h248** command to enter the MG interface mode.
2. Run the **if-h248 attribute** command to create an MG interface and then configure the primary/secondary MGCs and disaster-recovery MGC.

   When configuring an MG interface supporting dual homing, note that:

   - The MG is dual homed to the primary/secondary MGCs. The configurable parameters include **secondary-mgc-ip1** *secondary-mgc-ip1*, **secondary-mgc-ip2** *secondary-mgc-ip2*, **secondary-mgc-port** *secondary-mgc-port*, or **mgc-domain-name2** *mgcdomainname2*.
   - At least one secondary MGC (containing the IP address and port ID) is configured.

   When configuring an MG interface supporting multi-homing, note that:

  - A disaster-recovery MGC is configured based on the dual homing. The configurable parameters include **stand-alone-mgc-ip1** *stand-alone-mgc-ip1*, **stand-alone-mgc-ip2** *stand-alone-mgc-ip2*, and **stand-alone-mgc-port** *stand-alone-mgc-port*.
  - At least one disaster-recovery MGC (containing the IP address and port ID) is configured.

  For details about how to configure an MG interface, see 1.23.1 Configuring an MG Interface.

Step 2 Configure the software parameters of an MG interface.

1. Run the **mg-software parameter 2** command to configure the MG interface supporting dual homing.

   The values of **mg-software parameter 2** are described as follows:

   - When the value of **mg-software parameter 2** is **0** (default value), multi-homing is not supported. Specifically, after an MG interface is unable to register with the primary MGC, the MG interface does no initiate registration with the secondary MGC or disaster-recovery MGC though it has been configured.
   - When the value of **mg-software parameter 2** is **1**, multi-homing is supported but auto-switching is not supported. Specifically,
     - An MG interface registering with the primary MGC will register with the secondary MGC and then the disaster-recovery MGC cyclically when the primary MGC malfunctions.
     - After the primary MGC recovers, the secondary MGC or disaster-recovery MGC will not automatically switch back to the primary MGC. Run the **mgc switch(h248)** command to forcibly switch the MGC to the primary MGC.
   - When the value of **mg-software parameter 2** is **2**, multi-homing and auto-switching are supported. Specifically,
     - An MG interface registering with the secondary MGC will automatically switch to the primary MGC after the primary MGC recovers.
     - An MG interface registering with the disaster-recovery MGC will automatically switch to the primary or secondary MGC after the primary or secondary MGC recovers.

NOTICE

After an MG interface is manually or automatically switched, the MG interface will restart, causing services interrupted for a short time.

2. (Optional) Run the **mg-software parameter 36** command to configure the registration interval during MGC's multi-homing registration switchover.

   This parameter is used to configure the switch interval when an MGC (primary MGC, secondary MGC, or disaster-recovery MGC) malfunctions and switches to another MGC. It is defaulted to 40s.
3. Run the **display mg-software parameter** command to query the software parameters of the MG interface.

----End

Example

To configure dual homing on MG interface 0 with the following settings, do as follows:

- The IP address of the primary MGC is 192.168.0.10 and the port number for the transport layer protocol is 2944.
- The IP address of the secondary MGC is 192.168.0.20 and the port number for the transport layer protocol is 2944.
- Auto-switching is not supported.

```
huawei(config)#interface h248 0
huawei(config-if-h248-0)#if-h248 attribute primary-mgc-ip1 192.168.0.10 primary-
mgc-port 2944 secondary-mgc-ip1 192.168.0.20 secondary-mgc-port 2944
huawei(config-if-h248-0)#mg-software parameter 2 1
huawei(config-if-h248-0)#display mg-software parameter 2
  -------------------------------------------------
  Interface Id:0         para index:2  value:1
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   2: Whether MG support multi-home function
     0: Do not support the multi-homing
     1: Support the multi-homing, but do not support the auto switchover
     2: Support the multi-homing and auto switchover
```

To configure dual homing on MG interface 1 with the following settings, do as follows:

- The IP address of the primary MGC is 192.168.0.10 and the port number for the transport layer protocol is 2944;
- The IP address of the secondary MGC is 192.168.0.20 and the port number for the transport layer protocol is 2944;
- The IP address of the disaster-recovery MGC is 192.168.0.30 and the port number for the transport layer protocol is 2945;
- Auto-switching is supported.

```
huawei(config)#interface h248 1
huawei(config-if-h248-1)#if-h248 attribute primary-mgc-ip1 192.168.0.10 primary-
mgc-port 2944 secondary-mgc-ip1 192.168.0.20 secondary-mgc-port 2944 stand-alone
-mgc-ip1 192.168.1.30 stand-alone-mgc-port 2945
huawei(config-if-h248-1)#mg-software parameter 2 2
huawei(config-if-h248-1)#display mg-software parameter 2
  -------------------------------------------------
  Interface Id:1         para index:2  value:2
  -------------------------------------------------
 APPENDIX:
  -------------------------------------------------
   Interface software parameter name:
   2: Whether MG support multi-home function
     0: Do not support the multi-homing
     1: Support the multi-homing, but do not support the auto switchover
     2: Support the multi-homing and auto switchover
```

1.32.3.2 Configuring MGCP-based Dual Homing

This topic describes how to configure the MGCP-based dual homing.

Context

The MA5600T/MA5603T/MA5608T supports registering with three MGCs (MGC1, MGC2, and MGC3) through the MG interface. MGC1 serves as the primary MGC. When MGC1 fails, the MG can switch to MGC2 and continue working. When MGC2 also fails, the MG can switch to MGC3 and continue working.

Prerequisite

- MGC1 and MGC2 must be configured in the attributes of the MG interface.
- On the MGCs, the data for interconnecting with the MG must be configured.

Procedure

Step 1 In the global config mode, run the **interface mgcp** command to enter the MG interface mode.

Step 2 Run the **mg-software parameter 3 1** command to enable the heartbeat message function.

Step 3 Run the **mg-software parameter 2 0** command to configure the MG interface to support dual homing.

----End

Example

To configure MG interface 0 to support dual homing and enable the heartbeat message function, do as follows:

```
huawei(config)#interface mgcp 0
huawei(config-if-mgcp-0)#mg-software parameter 3 1
huawei(config-if-mgcp-0)#mg-software parameter 2 0
```

1.32.3.3 Configuring the SIP-based Dual Homing

the MA5600T/MA5603T/MA5608T supports the 1+1 mutual assistance mode (the active/standby mode) of the upstream proxy devices. When either of the upstream active/standby devices is faulty, the MA5600T/MA5603T/MA5608T automatically switches the service to the other device. In this way, the disaster recovery solution is implemented through SIP to improve the access reliability of the device.

Context

The MA5600T/MA5603T/MA5608T supports the SIP interface homing to two proxy servers (Proxy1 and Proxy2), where Proxy1 functions as the primary proxy server. When Proxy1 fails, the MG can switch to Proxy2 to continue working.

Prerequisite

- The data for interconnecting with the SIP interface must be configured on the IMS.
- When configuring the IP address of the SIP interface, make sure that the IP address (signaling IP address or media IP address) exists in the corresponding IP address pool.

Procedure

Step 1 In the global config mode, run the **interface sip** command to enter the SIP interface mode.

Step 2 Run the **if-sip attribute basic** command to configure the basic attributes of the SIP interface.

To support dual homing, the information about the secondary proxy server must be specified here. A proxy server can be identified by its IP address or domain name.

Step 3 Run the **reset** command to reset the interface.

----End

Example

Assume that the SIP interface ID is 0, the media IP address is 10.10.10.13, signaling IP address is 10.10.10.13, transfer protocol is UDP, and UDP port number is 5000; IP address 1 of the primary proxy server is 10.10.10.14, and port number of the primary proxy server is 5060; IP address 1 of the secondary proxy server is 10.10.10.15, and port number of the secondary proxy server is 5060; the homing domain name **huawei.com**, and profile ID is 1. To configure the dual homing attributes of SIP interface 0, do as follows:

```
huawei(config)#interface sip 0
huawei(config-if-sip-0)#if-sip attribute basic media-ip 10.10.10.13
signal-ip 10.10.10.13 signal-port 5000 transfer udp primary-proxy-ip1
10.10.10.14 primary-proxy-port 5060 primary-proxy-domain proxy.domain
secondary-proxy-ip1 10.10.10.15 secondary-proxy-port 5060
huawei(config-if-sip-0)#if-sip attribute basic home-domain huawei.com sipprofile
-index 1
huawei(config-if-sip-0)#reset
```