Troubleshooting No Busy Tone and No Announcement Messages on ISDN-VoIP (H.323) Calls

---

Contents

Problem Description

This document addresses call progress in-band related issues when interworking ISDN and H.323 signaling between VoIP and PSTN networks. Challenges arise when Cisco VoIP router/gateways exchange signaling capabilties with the telco switch. The following list describes common problem scenarios/symptoms:

- No DTMF Digits or Audio Passed on VoIP Calls to PSTN/PBX
  Symptom: IP phone user makes a call, is able to hear Anouncement Messages (for example "enter your account number..") but cannot pass DTMF digits. This symptom applies for both VoIP Toll-Bypass calls and IP Phone to PSTN/PBX calls.

- No Busy Tone or Announcement Message Received when Placing VoIP Outbound Calls
  Symptom: IP phone (CallManager scenario) or POTS phone (VoIP Toll-Bypass scenario) does not hear a busy tone or announcement message from the PSTN network. This symptom applies for both VoIP Toll-Bypass calls and IP Phone to PSTN/PBX calls.

For further information on ISDN - VoIP (H.323) call progress in-band related issues refer to:

[Troubleshooting No Ringback Tone on ISDN-VoIP (H.323) Calls]

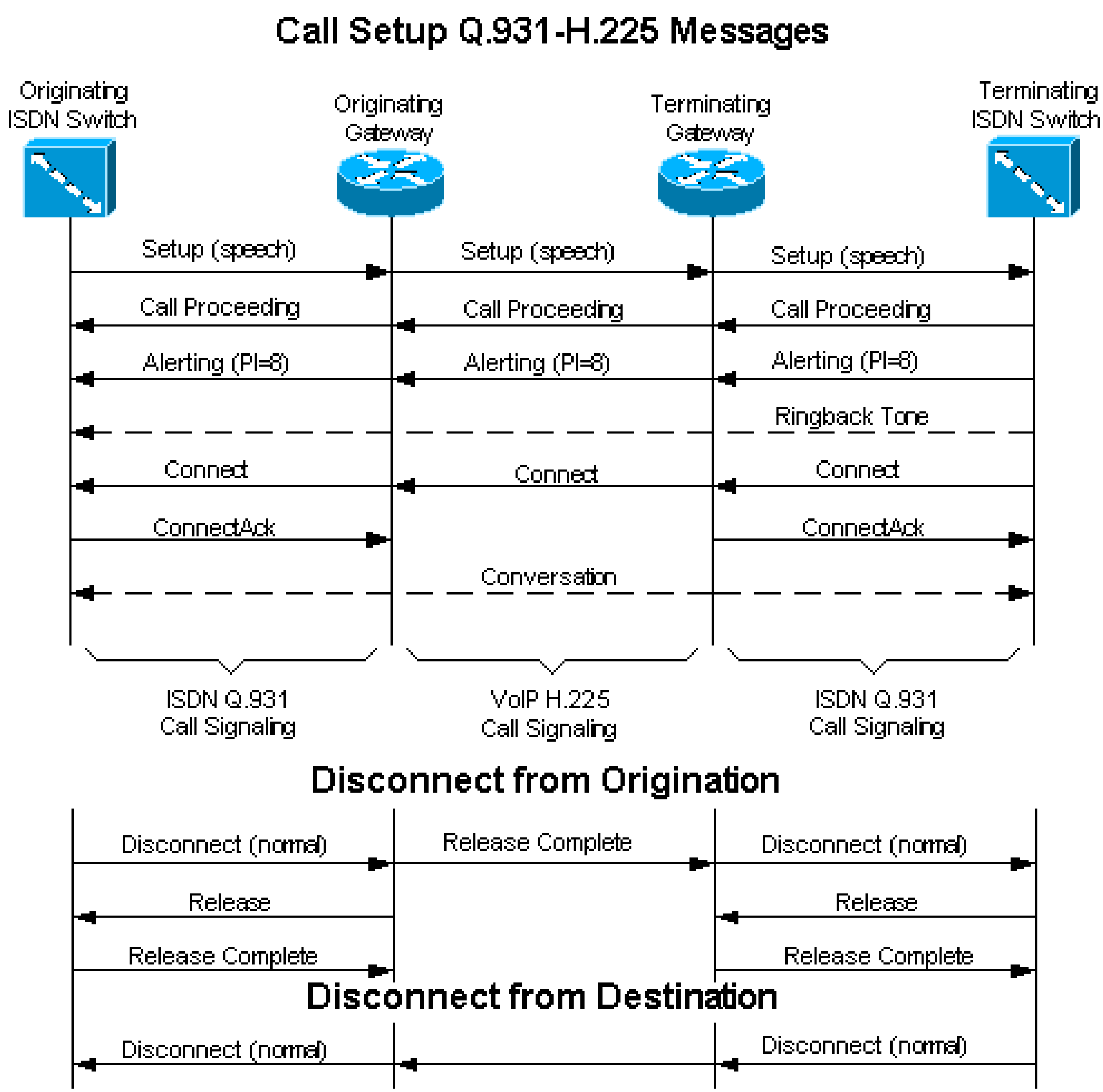
Note: We recommend reading the Background Information section before reading the Solutions section.

Background Information

ISDN-VoIP Interworking

Interworking is defined as as the mapping of call signaling messages between two different protocol suites. In the context of this document we will focus on ISDN and H.323 (VoIP) *interworking* issues. The following diagram displays the call signaling messages in the ISDN (Q.931) and VoIP (H.225) call leg.

Note: H.225 is a protocol specified by H.323 for call signaling and call setup. H.225 specifies the use and support of Q.931. For more information on H.323 refer to the [H.323 Tutorial].



Progress Tones and Progress Indicators

In-band progress tones (for example, ringback and busy tones) and announcements (for example, "The number you have dialed is no longer in service") are required to successfully signal voice calls. Progress tones can be generated by the originating, terminating or intermediate devices.

The indication of in-band tones and announcements is controlled by the *Progress Indicator (PI)* information element (IE) in ISDN and H.323 networks. The *Progress Indicator* signals those interworking situations where in-band tones and announcements must be used. In the context of this document, the following are the ITU Q.931 Progress Indicator values of interest:

- **Progress indicator = 1** – Call is not end-end ISDN. Further call progress information may be available in-band.
- **Progress indicator = 2** – Destination address is non-ISDN.
- **Progress indicator = 3** – Origination address is non-ISDN.
- **Progress indicator = 8** – In-band information or an appropriate pattern is now available.

The indication that tones and announcements are available is signaled by an Alerting, Call Proceeding, Progress, Connect, Setup Ack or Disconnect message containing a Progress Indicator = 1 or 8.

When a Setup message arrives to the originating gateway with a PI = 3, it means that the switch is informing the gateway that in-band messages are expected.

Note: A lack of a PI in a message assumes that the originating device will provide the appropriate tone signaling to the calling party.

Note: Analog and digital CAS (Channel Associated Signaling) PSTN circuits will usually carry the information as in-band information.

Voice Path Cut-through

Voice path cut-through is the completion of the bearer transmission path of a voice call. In a voice call, cut-through occurs in two stages:

- Cut-through in the Backward Direction – Means that only the voice path from the called party to the calling party is complete.
- Cut-through in Both Directions – Means that the voice path between the called and calling party is complete.

Tones and announcements may be generated at the origination switch or the destination switch. If tones and announcements are generated by the destination switch, then the voice path transmission path (backward) from the destination switch to the calling party must be cut-through prior to the time that the tones and announcements are generated. Early cut-through of the backward bearer path (before the Connect message) is needed to transport in-band tones and announcements from called party to the calling party and to avoid speech clipping.

The call terminating Cisco router/gateway cuts through the audio path in the backward direction to transmit in-band information when the terminating ISDN switch sends the following messages:

- Alert message with PI = 1 or PI = 8
- Progress message with PI = 1 or PI = 8
- Call Proceding message with PI = 1 or PI = 8
- Setup Ack message with PI = 1 or PI = 8
- Disconnect message with PI = 1 or PI = 8

Note: On terminating CAS interfaces, the Cisco router/gateway cuts through the audio in the backward direction once all called number digits are sent.

The terminating Cisco router/gateway cuts through the audio path in both directions in the following cases:

- Connect message is received on an ISDN interface
- Answer supervision (off-hook) is received on a CAS interface.

Cut-through in both directions can be set on the gateways through the use of the Cisco IOS global configuration command **voice rtp send-recv**

Solutions

In Cisco IOS® Software Releases 12.1(3)XI1 and 12.1(5)T, Progress indication was changed to provide better interworking between POTS and VoIP interfaces. This is mainly achieved through enabled and propagation end-end of the Progress Indication value that defines progress indication tone generation.

The usage of these commands assumes you are running at least 12.1(3a)XI5 or 12.2(1)or later.

For more information refer to: Interworking Signaling Enhancements for H.323 VoIP and Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2

No DTMF Digits or Audio Passed on VoIP Calls to PSTN/PBX

Symptom:
User makes a call, hears an Anouncement Messages (for example "enter your account number...") but cannot pass DTMF digits.
This symptom applies for both VoIP Toll-Bypass and IP Phone calls to PSTN/PBX calls.

Problem Description:
IP phone (CallManager scenario) or POTS phone (VoIP Toll-bypass scenario) call leaves through a

Cisco IOS gateway where the called number is usually an IVR system that sends back an ISDN progress message, but no connect until some account information is entered. By default, the audio path is cut-through in the backward direction (toward the IP Phone or originating gateway), but not in the forward direction until the terminating gateway receives a Connect message. Therefore, there is no voice path to transmit DTMF tones or speech towards the terminating switch.

Solution:
Configure the Cisco IOS global configuration command **voice rtp send-recv** to establish (cut-through) the audio path in both directions prior to receiving an ISDN Connect message from the PSTN. For more information on this command refer to:
Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2

No Busy Tone or Announcement Message Received when Placing VoIP Outbound Calls

Symptom:
IP phone (CallManager scenario) or POTS phone (VoIP Toll-Bypass scenario) does not hear a busy tone or announcement message from the PSTN network.

Solution:
Configure the Cisco IOS global configuration command **voice call convert-discpi-to-prog** (Cisco IOS Software Release 12.2(1) and later). This command converts an inbound ISDN Disconnect message with a PI to a H225 Progress message with the same PI value. This command can help at times when an announcement is being played on the terminating PSTN side, but the calling party does not hear the response.

In the VoIP Toll-bypass scenario, most of these issues are resolved by upgrading the router/gateways to a Cisco IOS Software Release of 12.1(3a)XI5 or 12.2(1) and later. However, if the originating device or originating ISDN switch does not keep the call active when a H225/ISDN disconnect message is received, try using the command above.

This may come up when the announcement in-band is a busy tone as well. Beyond that, the busy signal should be provided by either the terminating device, the originating device and the network. Some aspects of this may be controlled.

Related Information

- **Interworking Signaling Enhancements for H.323 VoIP**
- **Interworking Signaling Enhancements for H.323 and SIP VoIP**
- **PSTN Callers not Hearing any Ring Back When they Call IP Phones**
- **Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2**
- **Understanding debug isdn q931 Disconnect Cause Codes**

- **Voice, Telephony and Messaging Technical Tips**
- **Voice and Telephony Technology Support Page**
- **Voice and Telephony Product Support Page**
