

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

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Introduction

This document addresses call progress in–band related issues when interworking ISDN and H.323 signaling between VoIP and the Public Switched Telephone Network (PSTN). Challenges arise when Cisco VoIP routers/gateways exchange signaling capabilities with the telco switch. This list describes common problem scenarios/symptoms:

- No DTMF Digits or Audio Passed on VoIP Calls to PSTN/PBX An IP phone user makes a call, is able to hear announcement messages, such as "enter your account number...", but cannot pass Dual–Tone Multifrequency (DTMF) digits. This symptom applies for both VoIP Toll–Bypass calls and Cisco IP phone to PSTN/PBX calls.
- No Busy Tone or Announcement Message Received when Placing VoIP Outbound Calls A Cisco IP phone (CallManager scenario) or Plain Old Telephone Service (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.

Refer to Troubleshooting No Ringback Tone on ISDN–VoIP (H.323) Calls for more information on ISDN – VoIP (H.323) call progress in–band related issues.

Cisco recommends that you read the Background Information section before you read the Solutions section.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

This document is not restricted to specific software and hardware versions.

Conventions

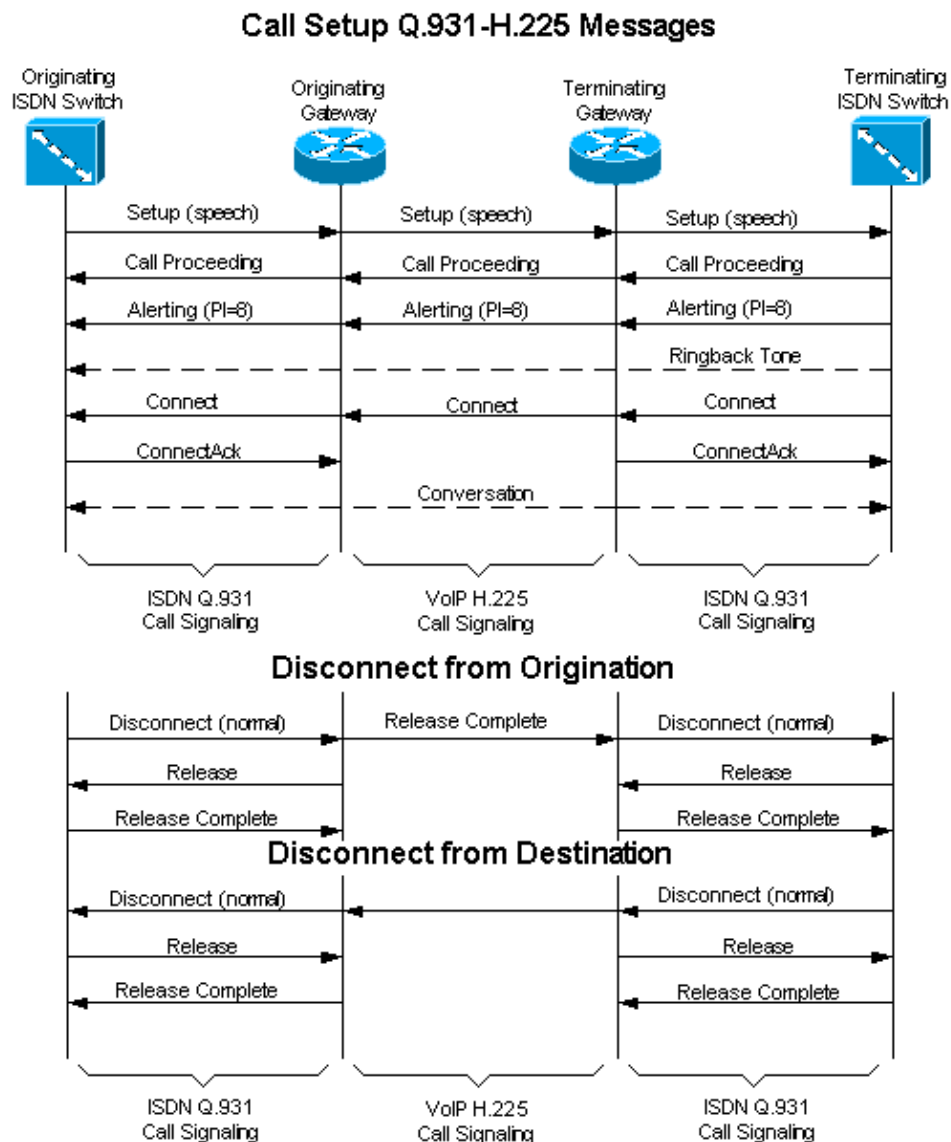
Refer to Cisco Technical Tips Conventions for more information on document conventions.

Background Information

ISDN-VoIP Interworking

Interworking is defined as the mapping of call signaling messages between two different protocol suites. In the context of this document, focus is on ISDN and H.323 (VoIP) interworking issues. This 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. Refer to the H.323 Tutorial for more information on H.323.



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 PI signals those interworking situations where in-band tones and announcements must be used. In the context of this document, these are the ITU Q.931 PI values of interest:

- **PI = 1** Call is not end-end ISDN. Further call progress information might be available in-band.
- **PI = 2** Destination address is non-ISDN.
- **PI = 3** Origination address is non-ISDN.
- **PI = 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 that contains a PI = 1 or 8.

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

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

Note: Analog and digital Channel Associated Signaling (CAS) PSTN circuits 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 can 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 these messages:

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

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 these 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 is changed to provide better interworking between POTS and VoIP interfaces. This is mainly achieved through enabled and propagation end-end of the PI value that defines progress indication tone generation.

The usage of these commands assumes you run at least Cisco IOS Software Release 12.1(3a)XI5 or 12.2(1) or later.

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

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

Symptom

The user makes a call, hears announcement messages, such as "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

A Cisco 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 Interactive Voice Response (IVR) system that sends back an ISDN progress message, but does not 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**, in order to establish (cut-through) the audio path in both directions prior to receiving an ISDN connect message from the PSTN. Refer to Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 for more information on this command.

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

Symptom

A Cisco 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 Software global configuration command, **voice call convert-discpi-to-prog**. This is used with Cisco IOS Software Release 12.2(1) and later. This command converts an inbound ISDN disconnect message with a PI to a H.225 progress message with the same PI value. This command can help when an announcement is 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 with an upgrade of 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 H.225/ISDN disconnect message is received, then issue the **voice call convert-discpi-to-prog** command.

This can 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, or the network. Some aspects of this can be controlled.

No Busy Tone on Inbound Call from Telephony (ISDN) to Cisco CallManager IP Phone, IOS Gateway or Third Party H323 Device

Symptom

A call from PSTN through gateway to a Cisco CallManager IP Phone, Cisco IOS gateway, or third party H.323 device might not hear a busy tone when it runs either an application or two-stage dialing on the originating gateway.

Solution

This is a less common case that can occur when the originating gateway either runs a voice application such as debit-card, or is running two-stage dialing. The latter refers to the calling party that dials the number to the gateway first, receives the dial tone, then dials the called party. In either case, the call has connected in terms of the PSTN network once it terminates on the originating gateway. If the IP call leg comes back with a release with the cause user-busy, this cannot be indicated back towards the telephony session that is in a connect state.

This has been addressed by having the originating gateway generate a busy tone when the release from the IP call leg is received with a cause code of user busy. The telephony leg is either released by the calling party or by the gateway after several minutes with the cause code of normal call clearing.

This feature is available from Cisco IOS Software Release 12.2(8)/12.2(8)T and later.

Note: In order to initiate a full-consult transfer from an IP phone which is registered to Cisco CallManager Express, the IP phone needs to have more than one line available. You need to configure and issue the **ephone-dn [number] dual-line** command. This allows the IP phone to have two lines or channels associated with the one directory number. The normal behavior with dual-line configured is that if a call is already active on the first channel, and another call is made to that extension, the caller hears the alert tone (ringing) on the second channel instead of a busy tone. If you want a busy tone to be received by the caller when an extension is busy on the first channel, you need to configure and issue the **huntstop channel** command under **ephone-dn**, as shown in this example:

```
CMECUE(config)#ephone-dn 1
CMECUE(config-ephone-dn)#huntstop channel
```

!--- Stops hunting on the second channel of a dual-line dn.

Related Information

- **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 Technology Support**
- **Voice and IP Communications Product Support**
- **Recommended Reading: Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation – Cisco Systems**

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