Inability To Break Dialtone in a Voice over IP Network

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Introduction

Inability to break dial tone is a common problem encountered in a VoIP network. In this scenario, the calling party is unable to pass the dual tone multifrequency (DTMF) tones or digits to the terminating device. This, in turn, does not let callers dial the desired extension or interact with the device that needs DTMF tones (such as voice mail or interactive voice response [IVR] applications). This problem could be caused by any of these issues:

- DTMF tones are not passed.
- DTMF tones are not understood.
- DTMF tones are passed but are not understood due to distortion.
- Other signaling and cabling issues.

This document addresses the most common problems and solutions.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

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

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

For more information on document conventions, refer to the Cisco Technical Tips Conventions.

Problem

The router puts a seizure on the local PBX, but the dial tone remains while the user is dialing.

Solutions

Solution 1

Ensure that the dial−type is set as `dtmf` on both the router and the PBX, as shown in the next sample output. Because the Foreign Exchange Station (FXS) port does not pass on digits, this setting is not available on an FXS port. However, this setting *can* be changed on Foreign Exchange Office (FXO) ports and on receive and transmit (Ear and Mouth [E & M]) ports.

```
Router(config−voiceport)# dial−type ?

  dtmf   touch−tone dialer
  mf     mf−tone dialer
  pulse  pulse dialer
```

Solution 2

In case of E & M, issue a **show call active voice brief** command to ensure that you are receiving the answer supervision from the PBX. The status of the call should be `active`, if you have received answer supervision. If the Telephony leg is still in the `connecting` state, then the router will not completely close the audio path. If this is the case, then you should contact the PBX vendor and ask them to provide answer supervision.

A workaround to this problem is to try to change the signaling on the router to `immediate` (see the next sample output) and then issue the **auto cut−through** command under the voice port. The router can then bring the call up to `active` state and cut through the audio.

```
Router(config−voiceport)# signal ?

delay−dial  delay before dialing
immediate   start immediately
wink−start  start upon wink

Router(config−voiceport)# ?

Voice−port configuration commands:
auto−cut−through  E & M auto cut−through without answer signal
```

Note: The signaling should match between the router and the PBX. Otherwise, calls in one direction might not work.

Solution 3

In the case of analog E&M, ensure that all cabling is installed correctly as described in Understanding and Troubleshooting Analog E & M Interface Types and Wiring Arrangements. Correct installation ensures that both transmit and receive audio paths are mapped correctly. Incorrect installation can cause audio paths not to establish properly and, therefore, the digits will not pass correctly between the two connected devices. The desired extension is reached, but the terminal device does not understand the tones when they are pressed.

Solution 4

In the case of a VoIP call from an originating gateway (OGW) to a terminating gateway (TGW), terminating the call to a Telephony device might not be understood. When you are passing DTMF tones through a compressed VoIP audio path, some or part of the dual tones could become slightly distorted because digital signal processor (DSP) codecs are designed to interpret human speech, not machine tones. Usually, such distortion does not occur with earlier compression codecs, such as G.723 or G.711, but later compression codecs can cause distortion of in−band tones. Cisco IOS® Software Release 12.0(5)T allows the DTMF tones to be passed out−of−band between VoIP gateways via three different techniques. All of these techniques use the H.245 capabilities exchange (part of H.323v2) to signal to the remote VoIP gateway that a DTMF tone has been received and that the remote VoIP gateway should regenerate it.

Issue the **dtmf−relay** command under the VoIP dial−peer on both sides. There are three different types of DTMF relays that can be configured:

```
Router(config)# dial−peer voice xxx voip

Router(config−dial−peer)# dtmf−relay ?

cisco−rtp           Cisco Proprietary RTP
h245−alphanumeric   DTMF Relay via H245 Alphanumeric IE
h245−signal         DTMF Relay via H245 Signal IE
```

Try a different setting for the **dtmf−relay** command. The **cisco−rtp** setting is proprietary to Cisco and is available prior to Cisco IOS Software Release 12.0(5)T. The other two settings follow the H.323v2 standards.

For Media Gateway Control Protocol (MGCP) networks, refer to MGCP Based Fax (T.38) and DTMF Relay.

For session initiation protocol (SIP) networks, refer to Dual Tone Multifrequency Relay for SIP Calls Using Named Telephone Events.

Solution 5

The sent in−band tones might be distorted because of the configuration of the voice ports.

The tones sent across the network might have a signal strength that is too low or too high. You can adjust the `input gain` and `output attenuation` of the signal to change the signal strength. The configuration is found under the voice ports.

```
Router(config−voiceport)# input gain ?

<−6 − 14>  gain in db

Router(config−voiceport)# output attenuation ?

<−6 − 14>  attenuation in db
```

You can increase or decrease the signal at input. The exact value varies from vendor to vendor (the Telco). Normally this is +7. However, you can always try to increase or decrease by one until it reaches optimum stage. If the values of these parameters are set too low or too high, you might have problems. Adjust the values. The default values are 0 for both settings.

Solution 6

In addition to the previous issues, one−way audio can also contribute to this type of problem. When there is one−way audio, the digits sent across do not reach the intended destination. A common way to establish audio paths in both directions is to issue the **voice rtp send−recv** command on both routers. For more information to troubleshoot one−way audio, refer to Troubleshooting One Way Voice Issues.

If none of these solutions resolve your problem, contact the Cisco Technical Support.

Related Information

- **Analog E&M Voice Signaling Overview**
- **Voice Network Signaling and Control**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**