

Configure RTP Source Port Validation

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Introduction

This document describes how to enable Real Time Protocol (RTP) source port validation in order to avoid voice quality problem like crosstalk.

Prerequisites

Requirements

IOS 12.4(6)T or newer versions.

Components Used

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

Caution: 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, ensure that you understand the potential impact of any command.

Background Information

Public Switch Telephone Network (PSTN) callers experience crosstalk or mixed voice streams in this call flow:

IP Phones -- Cisco Unified Communications Manager (CUCM) --- Session Initiation Protocol (SIP)
IOS Gateway -- PSTN

This could happen when the gateway receives an invalid RTP stream destined to the same IP address and port of an active call.

The invalid stream has different source IP address and port than the one negotiated via SIP Session Description Protocol (SDP).

Configuration Check:

Check whether:

a) Hoot n Holler is being used:

```
dial-peer voice x voip
session protocol multicast
```

[CLI definition](#)

This command is used for voice conference configurations in a hoot and holler networking implementation. This command allows more than two ports to join the same session simultaneously.

b) If SIP is configured:

```
dial-peer voice x voip
session protocol sipv2
```

[CLI definition](#)

Configuration Steps

Configure these commands:

```
voice service voip
sip
source filter
```

This command eliminates the risk for crosstalk since the gateway blocks all rogue audio from an unknown source.

Note: The above command works for SIP only, so H323, Media Gateway Control Protocol (MGCP), and Skinny Client Control Protocol (SCCP) are still affected.

Configuration example:

```
voice service voip
allow-connections sip to sip
sip
bind control source-interface loopback0
bind media source-interface loopback0
source filter
```

```
dial-peer voice 2001 voip
```

```
destination-pattern 79...  
session protocol sipv2  
session target ipv4:172.16.32.21  
incoming called-number .  
voice-class codec 1  
dtmf-relay rtp-nte
```

Verify

There is currently no verification procedure available for this configuration.

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.

[Additional Reference](#)