SIP Phone: Call Failure on Consultation Transfer from Anonymous Callers

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Introduction

The Session Initiation Protocol (SIP) Call Transfer and Call Forwarding Supplementary Services feature implements SIP support of blind or attended call transfers and call forwarding requests from a Cisco IOS® gateway. A call transfer is considered consultative when the transferring parties either connect the caller to a ringing phone (ringback heard) or speak with the third party before connecting the caller to the third party. When you use the Cisco IP phone with SIP phone load, the consult call transfer might not work when the caller ID has an anonymous parameter or the caller ID is null. This document explains how to solve this issue.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on these software and hardware versions:

- Cisco IP Phone with a SIP load later than version 7.2
- Cisco IOS Voice Gateway that runs Cisco IOS Software Release 12.4(9)T

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

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

Anonymous Caller ID

When a consult call transfer is done for an incoming call with caller ID null on an IP phone with the SIP phone load, the caller ID is replaced with **anonymous**. These kind of calls are dropped and result in a failed call transfer.

Solution

In order to resolve this problem, you need to use a voice translation rule to replace the null caller ID with any other caller ID.

Create a Voice Translation Rule

You can change the calling line ID (CLID) when you use the **voice translation−rule** command on the gateway as this procedure shows.

1. Create a voice translation rule.

```
VoiceGateway(config)#voice translation−rule 1
VoiceGateway(cfg−translation−rule)#rule 1  /^$/ /2025551150/
VoiceGateway(cfg−translation−rule)#rule 3 /9999999999/  /2025551150/
```
2. Associate the translation rule with a voice translation profile.

```
VoiceGateway(config)#voice translation−profile changeNumber
VoiceGateway(cfg−translation−profile)#translate calling 1
```

   The rule 1 under the voice translation changes the calling number if it is null and rule 3 replaces it if the calling number is 9999999999. For more details about the **voice translation−rule** command, refer to Voice Translation Rules.
3. Enable this profile in a voice dial−peer.

```
VoiceGateway(config)#dial−peer voice 2000 voip
VoiceGateway(config−dial−peer)#translation−profile outgoing changeNumber
```

   This voice translation of the caller ID at the voice gateway prevents the failure of consult call forward.

Related Information

- **Troubleshoot a SIP Call Between Two Endpoints**
- **Voice Technology Support**
- **Voice and IP Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**

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