Contents

Introduction

The document describes a solution to a problem found when the same gateway is used as for Public Switched Telephone Network

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Outbound Dialer
- Cisco Unified Communications Manager (
- Cisco IOS Voice Gateways

Components Used

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

- CUCM Version 11
- Cisco IOS Voice Gateway: c2800nm-adventerprisek9_ivs-mz.151-2.T5

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.

Background Information

The voice gateway generates a ringback tone to the customer in specific call flows when the call is sent to the agent. For dialer call flows, in order to prevent the generation of a ringback from gateway, Session Initiation Protocol (SIP) normalization script to the Unified Communications Manager SIP trunk.

In the scenario where the same gateway is used for outbound dialer and PSTN calls, the trunk for PSTN calls still needs a 180 RINGING SIP message for inbound calls in order to trigger the gateway to play ringback to the PSTN, but needs to be disabled for outbound dialer calls.

Here is an example of the two scenarios described:

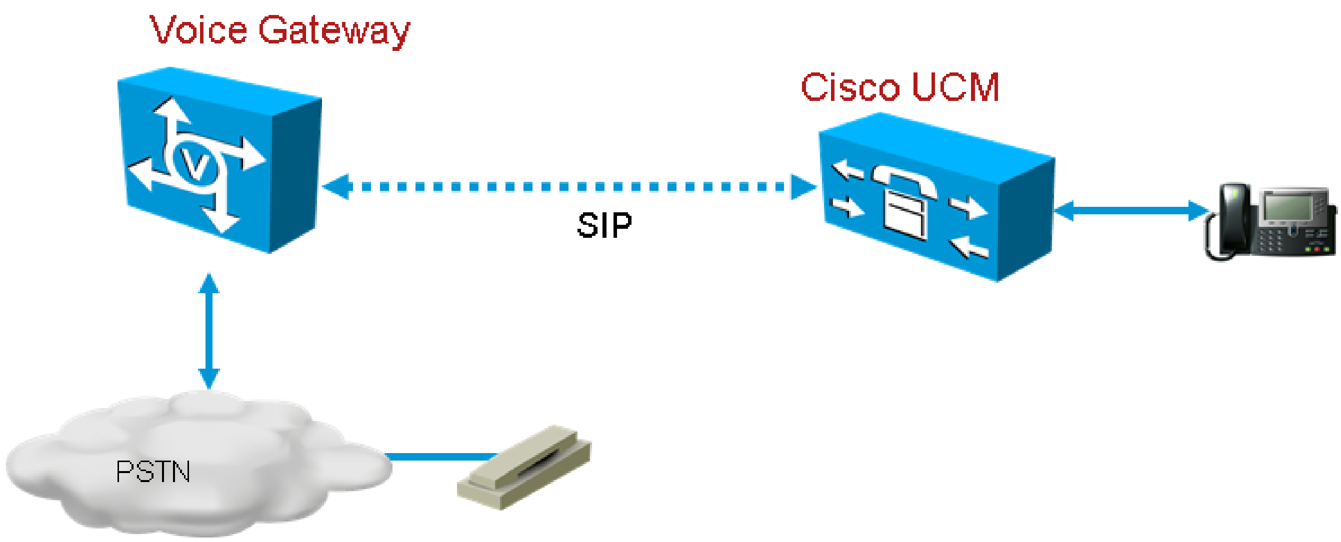

Figure 1. PSTN Calls

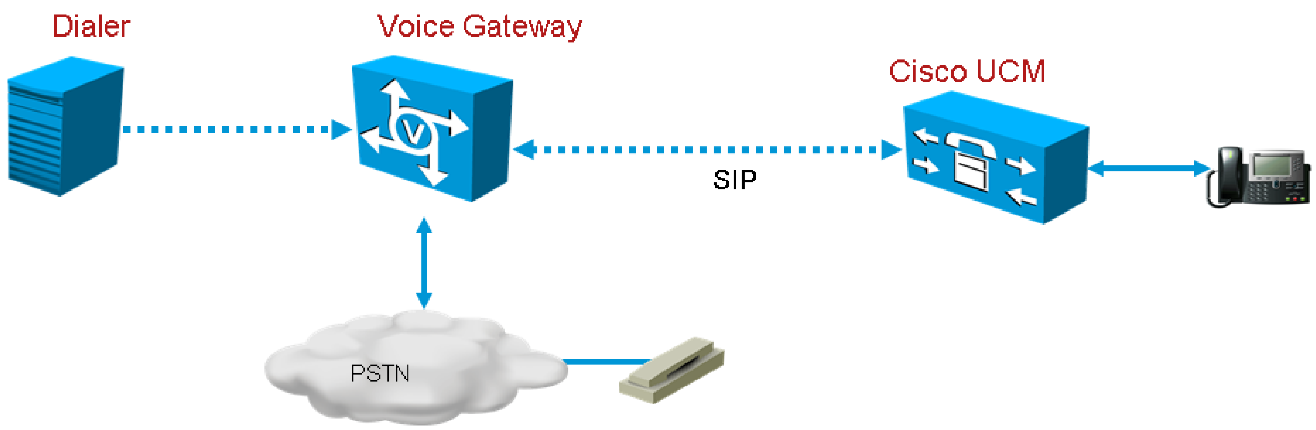

Figure 2. Dialer Calls

Configure

Since the SIP normalization script will be applied only to the Gateway trunk used for dialer calls, and the same gateway is used for Dialer and PSTN calls, an additional Gateway trunks needs to be created in CUCM. However, in CUCM you cannot add the same trunk twice unless the trunk uses a different incoming port. So in this scenario, the gateway trunk used for Dialer will have a different incoming port from the Gateway trunk used for the PSTN calls. It will be the same gateway, but with different incoming ports

CUCM

Step 1. Navigate to **https://*<IP_address>*:8443 where *<IP_address>*** identifies the CUCM.

Step 2. Sign in to CUCM.
Step 3. In order to create a SIP trunk security profile in CUCM, choose **Communications**

Manager GUI > System > Security > SIP Trunk Security Profile > [Add New]. The default port is 5060. Change the default port to 5065 or any SIP port available for the gateway and CUCM.

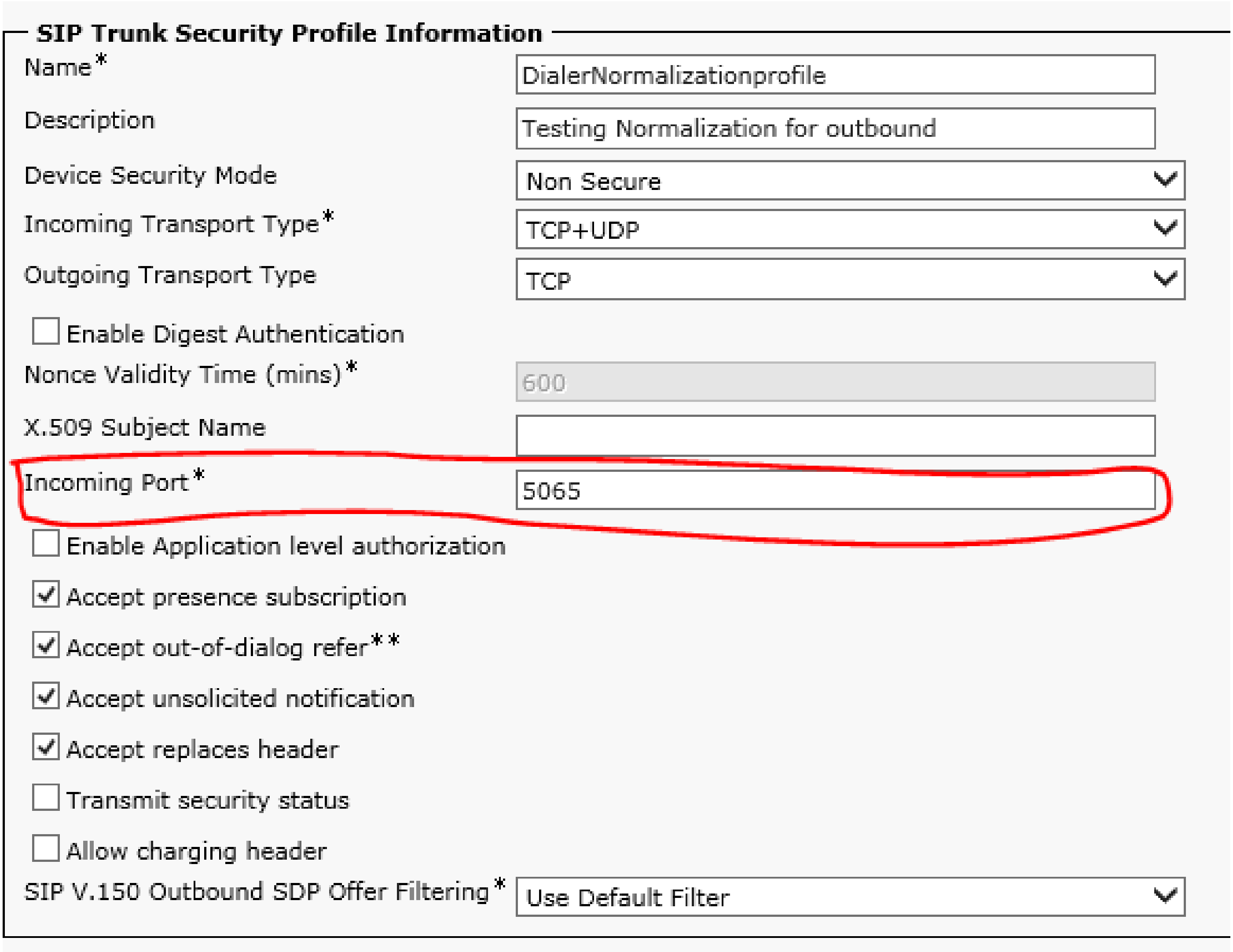

Figure 3. SIP Security Profile

Step 4. Click **Save**.

Step 5. Create a new SIP trunk and add the new SIP Trunk Security Profile.

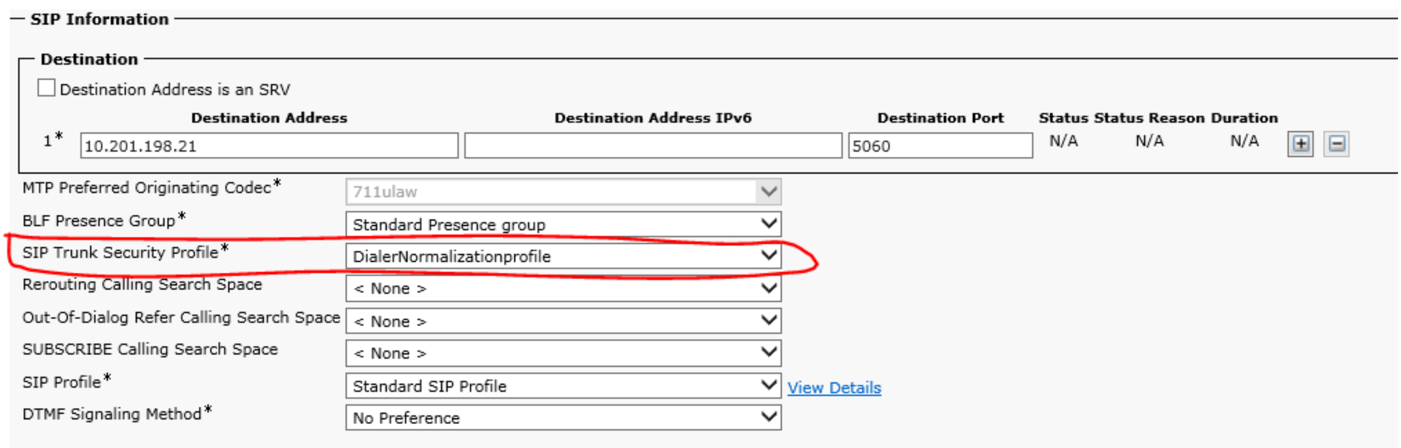

Figure 4. Create a New SIP Trunk

Step 6. Click **Save**.

Step 7. Click **Reset**.

Step 8. In **Communications Manager GUI > Devices > Device Settings > SIP Normalization Scripts > [Create New]**, enter this SIP normalization script into the content field. All other values remain set to default.

```
M = {}

function M.outbound_180_INVITE(msg)

msg:setResponseCode(183, "Session in Progress")

end

return M
```

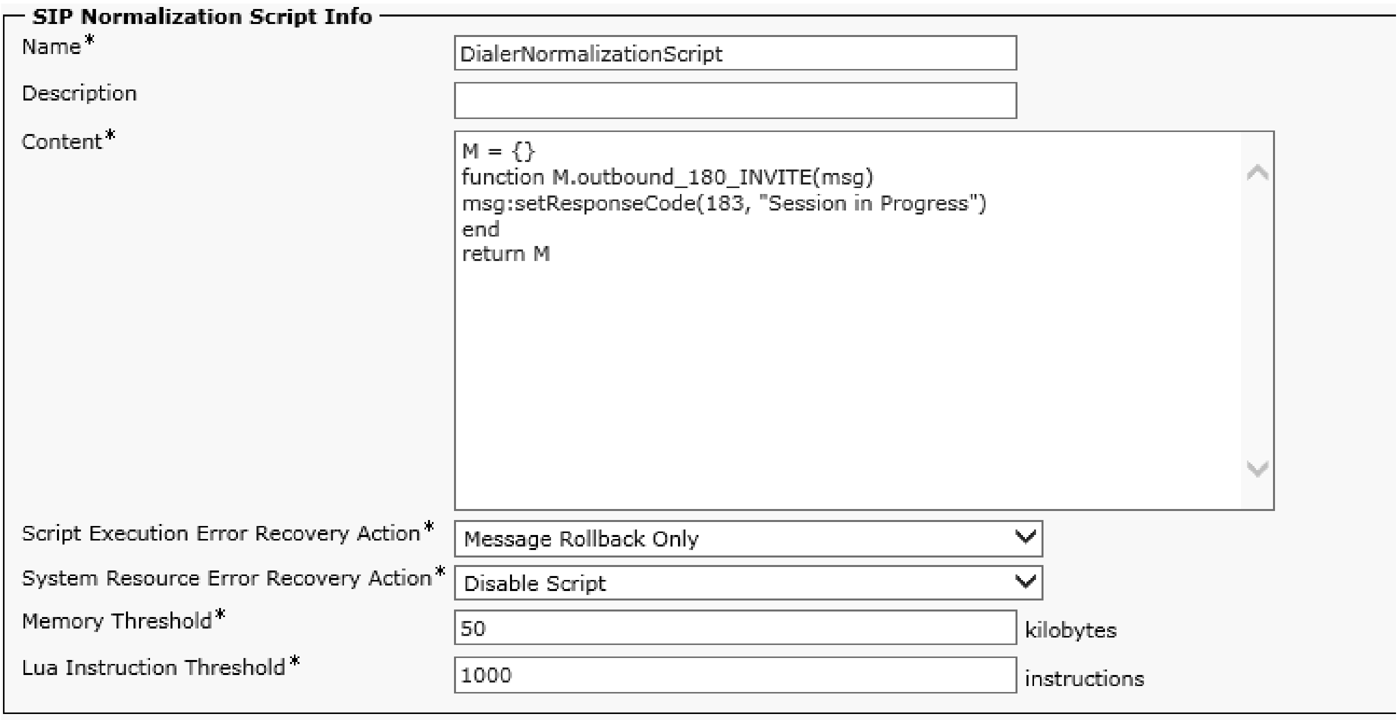

Figure 5. Add Normalization Script

Step 9. Click **Save**.

Step 10. Associate the new normalization script with the SIP trunk.

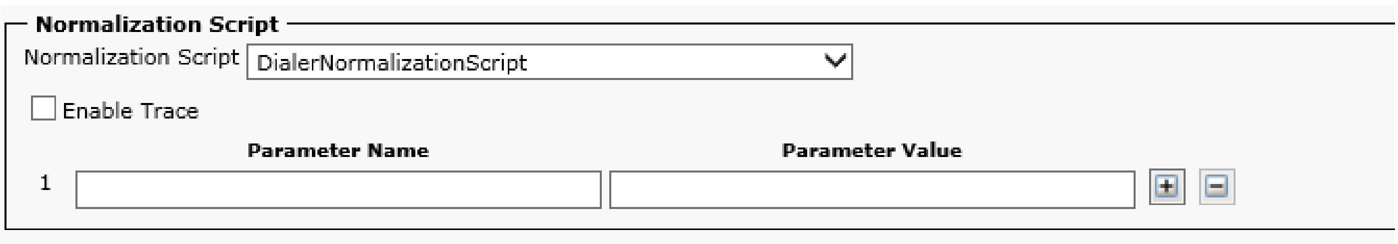

Figure 6. Associate Script with Trunk

Voice Gateways

In addition to the gateway configuration described on the Cisco Packaged Contact Center Enterprise Features Guide, Release 11.0, configure an outgoing Dial-peer for transferring call to the agent with the incoming port set on the CUCM SIP Trunk Security Profile (the port 5065 was used in the previous example).

Configure an Outgoing Dial-Peer to Transfer a Call to an Agent

This example shows this:

```
dial-peer voice 11000 voip
 destination-pattern 11T
 session protocol sipv2
 session target ipv4:10.10.10.31:5065(this is Call Manager's IP address and Security profile
incoming port)
 voice-class codec 1
 voice-class sip rel1xx supported "100rel"
 dtmf-relay rtp-nte h245-signal h245-alphanumeric
 no vad
```

Verify

There is currently no verification procedure available for this configuration.

Troubleshoot

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