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Topic: Configuring a Trunk in Cisco Unified Call Manager v11.0 for GCK

Background

This document describes the steps necessary to configure Cisco Unified Call Manager (CUCM) version 11.0 in order to interface with AtlasIED GCK version 1.0 or higher running in Trunking Mode. In this mode, GCK acts like a series of softphones belonging to a separate SIP trunk, which can be set up to work with CUCM. This guide will also provide instructions for connecting GCK to a CUCM trunk and how to test the trunk.

This document assumes the following:

- The user has a functioning installation of CUCM.
- CUCM has enough license units available to allow a SIP trunk to be installed. Contact Cisco for licensing information.

Installation

The process of creating a new SIP trunk consists of the following steps:

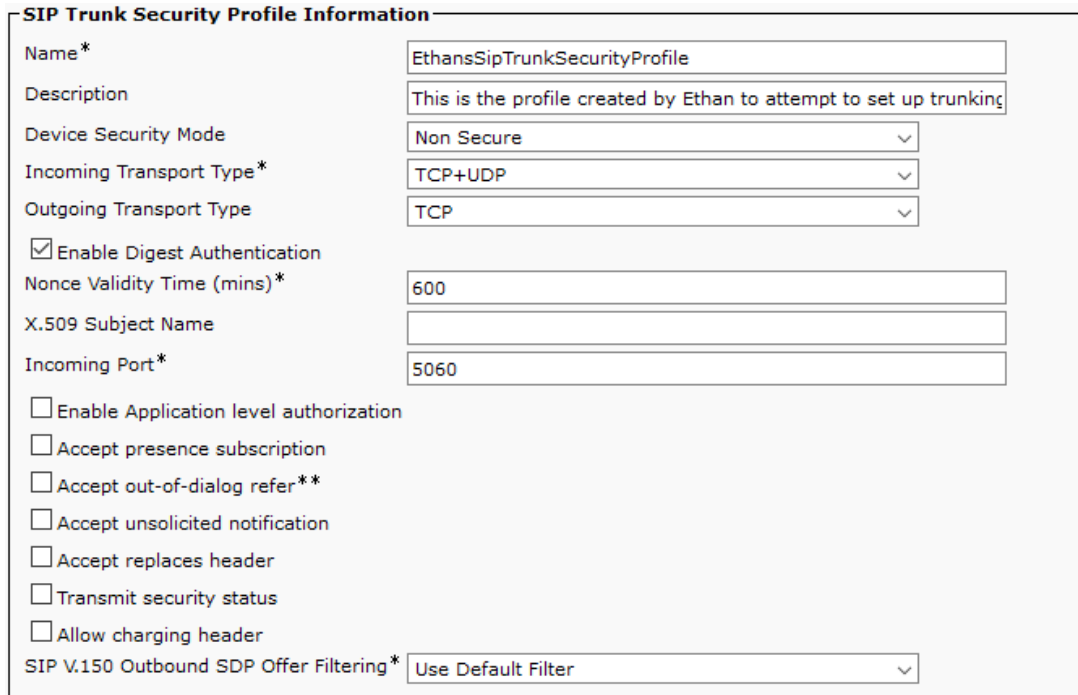
1. Configuring a new Trunk Security Profile.
2. Adding a new SIP Profile.
3. Adding a new Trunk Device Profile.
4. Adding a new Route Group.
5. Adding a new Route List.
6. Adding a new Route Pattern.
7. Configuring GCK with CallManager trunking.
8. Testing.



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Configure the Trunk Security Profile

1. Navigate to System > Security > SIP Trunk Security Profile.
2. Click "Add New" to bring up the SIP Trunk Security Profile Configuration screen (Figure 1).



The image shows the 'SIP Trunk Security Profile Information' configuration screen. It contains the following fields and options:

- Name***: Text input field containing 'EthansSipTrunkSecurityProfile'.
- Description**: Text input field containing 'This is the profile created by Ethan to attempt to set up trunking'.
- Device Security Mode**: Dropdown menu set to 'Non Secure'.
- Incoming Transport Type***: Dropdown menu set to 'TCP+UDP'.
- Outgoing Transport Type**: Dropdown menu set to 'TCP'.
- Enable Digest Authentication**: Checked checkbox.
- Nonce Validity Time (mins)***: Text input field containing '600'.
- X.509 Subject Name**: Text input field (empty).
- Incoming Port***: Text input field containing '5060'.
- Enable Application level authorization**: Unchecked checkbox.
- Accept presence subscription**: Unchecked checkbox.
- Accept out-of-dialog refer****: Unchecked checkbox.
- Accept unsolicited notification**: Unchecked checkbox.
- Accept replaces header**: Unchecked checkbox.
- Transmit security status**: Unchecked checkbox.
- Allow charging header**: Unchecked checkbox.
- SIP V.150 Outbound SDP Offer Filtering***: Dropdown menu set to 'Use Default Filter'.

Figure 1

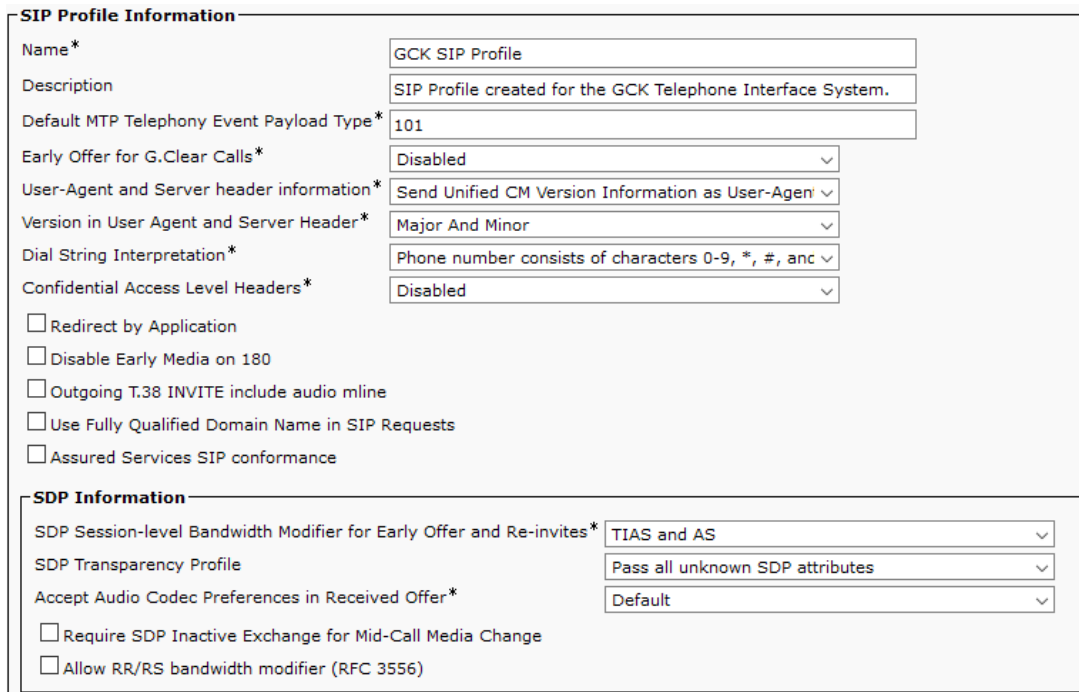
3. Fill in the profile name.
4. Set the outgoing transport type to UDP.
5. Check the Enable Digest Authentication box.
6. Click Save.



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Add New SIP Profile

1. Navigate to Device > Device Settings > SIP Profile.
2. Click Find to list all of the SIP Profiles and then click on Standard SIP Profile (Figure 2).



SIP Profile Information

Name* GCK SIP Profile

Description SIP Profile created for the GCK Telephone Interface System.

Default MTP Telephony Event Payload Type* 101

Early Offer for G.Clear Calls* Disabled

User-Agent and Server header information* Send Unified CM Version Information as User-Agent

Version in User Agent and Server Header* Major And Minor

Dial String Interpretation* Phone number consists of characters 0-9, *, #, and

Confidential Access Level Headers* Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* TIAS and AS

SDP Transparency Profile Pass all unknown SDP attributes

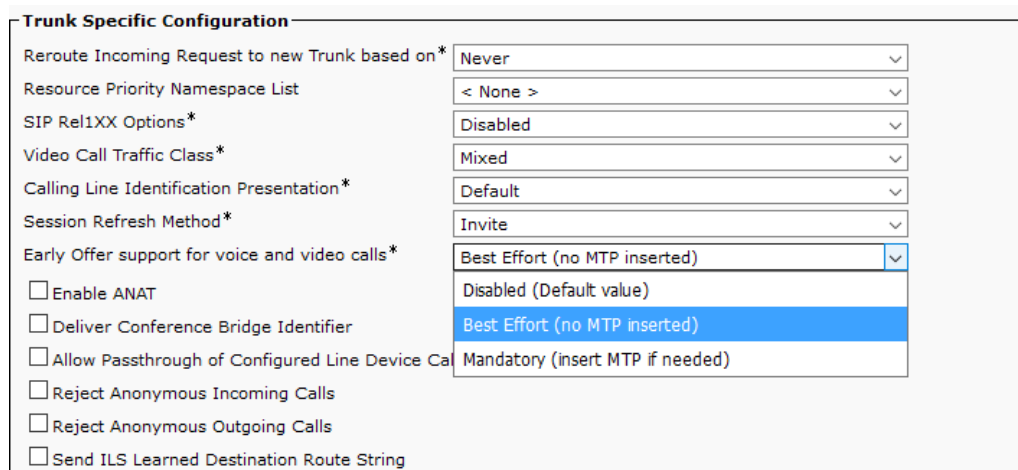
Accept Audio Codec Preferences in Received Offer* Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)

Figure 2

3. Click Copy.
4. Fill out the device name.
5. Fill out the device description.
6. Scroll down to the Trunk Specific Configuration section and select "Best Effort (no MTP inserted)" (Figure 3) from the Early Offer support for voice and video calls dropdown menu.



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

Resource Priority Namespace List < None >

SIP Rel1XX Options* Disabled

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

Session Refresh Method* Invite

Early Offer support for voice and video calls* Best Effort (no MTP inserted)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Call

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

Dropdown menu options:
 Disabled (Default value)
 Best Effort (no MTP inserted)
 Mandatory (insert MTP if needed)

Figure 3



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7. Scroll down to the SDP Information section and check the checkbox for “Allow multiple codecs in answer SDP” (Figure 4).

Figure 4

Add Trunk Device profile

1. Navigate to Device > Trunk.
2. Click “Add New” to bring up the Trunk Configuration screen .
3. Select SIP Trunk for the Trunk type and leave the rest of the settings at their default (Figure 5).

Figure 5

4. Click Next to open the Trunk Configuration window (Figure 6).

Figure 6

5. Fill in the Device Name.
6. Select a device pool from the Device Pool dropdown menu.



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7. Scroll to the SIP Information section and fill in the Destination Address with the IP address of the GCK controller (Figure 7).

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.2.133.155		5080

Figure 7

8. Set the Destination port to 5080.

9. Select a trunk security profile in the “SIP Trunk Security Profile” dropdown menu (Figure 8).

MTP Preferred Originating Codec*	711ulaw	▼
BLF Presence Group*	Standard Presence group	▼
SIP Trunk Security Profile*	EthansNewSIPTrunk	▼
Rerouting Calling Search Space	< None >	▼
Out-Of-Dialog Refer Calling Search Space	< None >	▼
SUBSCRIBE Calling Search Space	< None >	▼
SIP Profile*	GCK SIP Profile	▼ View Details
DTMF Signaling Method*	No Preference	▼

Figure 8

10. Select a sip profile “SIP Profile” dropdown menu.

11. Click Save.

Add a New Route Group

1. Navigate to Call Routing > Route/Hunt > Route Group.

2. Click “Add New” to open the Route Group Configuration screen (Figure 9).

Route Group Information

Route Group Name* GCK Route Group

Distribution Algorithm* Circular ▼

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices** GCKBox

Port(s) None Available ▼

Add to Route Group

Current Route Group Members

Selected Devices (ordered by priority)* GCKBox (All Ports) ▼

Reverse Order of Selected Devices

Removed Devices***

Figure 9

3. Fill in the Route Group Name.



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4. In the “Find Devices to Add to Route Group” section, highlight the trunk you want to configure and click “Add to Route Group”.
5. Click Save.

Add a New Route List

1. Navigate to Call Routing > Route/Hunt > Route List.
2. Click Add new to bring up the Route List Configuration Screen (Figure 10).
3. Fill in the Name.

Figure 10

4. Select the appropriate manager group.
5. Click Save.
6. After clicking save, more settings will appear in the window. In the “Route List Member Information” section, click the “Add Route Group” button (Figure 11).

Figure 11

7. In the Route List Detail Configuration window (Figure 12) Select the route group you created from the Route Group dropdown menu.

Figure 12



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8. Select "NANP:PreDot" from the Discard Digits dropdown menu.
9. Click Save.

Add a New Route Pattern

1. Navigate to Call Routing > Route/Hunt > Route Pattern.
2. Click "Add New" to bring up the Route Pattern Configuration screen (Figure 13).

Figure 13

3. Fill in the Route Pattern. We recommend using 8.! to set the route pattern to call to the GCK box when 8 is dialed in front of the extension.
4. In the Gateway/Route List dropdown menu, select the route list that was created above.
5. Click Save.

Configuring GCK with CallManager Trunking

1. Log in to GCK as an admin.
2. Navigate to the Configuration tab.
3. Navigate to the Telephone sub-tab.
4. Select "External via Trunking" from the SIP Configuration Mode dropdown menu (Figure 14).

Figure 14



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Testing

1. Add a new direct extension to the system by clicking the “+” button above Direct Extensions.
2. Fill in the Ext, SIP User, and SIP Password fields (we recommend using the same value in each field if possible).
3. Click “Edit” in the action performed field to bring up the action details configuration window (Figure 15).

Action Details

Copy Action

Action specific to device: PGH Test System (My Controller) (Telephone Interface)

Type: Live Announcement Show All Fields

Description: Test announcement for phone interface

Announcement Class: Telephone Interface AC (Pri:5)

Zone Group

Destination: + All-Call System-Wide (100) X

OK Cancel

Figure 15

4. Select Live Announcement for the type.
5. Select All-Call as the destination for the announcement (This is assuming you have created an all-call destination in your controller).
6. Dial the extension number preceded by an 8 and verify that you can hear audio coming through the speakers.