How to Connect Cisco Unified Call Manager to NeoGate TA FXS Gateway

Version: 1.0

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1. Introduction

This application note shows how to connect Cisco Unified CallManager to NeoGate TA FXS gateway. This guide has been tested with NeoGate TA3200 and CUCM version: 8.6.2

Two modes are available for NeoGate TA to connect to CUCM, we call them VoIP trunk mode and Service Provider SIP mode (SPS).

*You can simply choose one mode to achieve the connection.

Target:
After connecting NeoGate TA3200 and CUCM, FXS extensions will be extended on CUCM. Once the FXS ports are connected to CUCM, the phones connected to the FXS ports will be treated as CUCM extensions (Directory number). The following features can be achieved:
1. Make calls between the analog phone and CUCM SIP extension;
2. Receive incoming calls on the analog phones;
3. Make external calls from the analog phones using trunks on CUCM.

Description:
IP address of CUCM: 192.168.10.200
IP address of NeoGate TA3200: 192.168.10.126
2. Preparation

2.1 Preparation on TA3200

NeoGate TA3200 attempts to contact a DHCP server in your network to obtain valid network settings (e.g., the IP address, subnet mask, default gateway address and DNS address) by default.

Before connecting NeoGate TA3200 to CUCM, you need to set a static IP address from the same LAN with the CUCM.

**Steps to set static IP for NeoGate TA3200:**

1. Connect one analog phone in one of the FXS ports on NeoGate TA3200;
2. Dial "***" to enter the voice menu;
3. Press "1" to check the IP address;
4. Open the browser and type in the TA3200’s IP address to access TA3200;
5. Log in TA3200 and go to System → Network Preferences → LAN Settings, disable DHCP and configure the LAN network, save and apply the changes, reboot NeoGate TA3200 to take effect.

![Figure 2](image)

2.2 Preparation on CUCM

Please make sure Cisco CallManager Service is activated on CUCM.
Steps to activate Cisco Call Manager:
1. Log in to the CM Administration Web interface.
2. In the field “Navigation” on the top right corner, select **Cisco Unified Serviceability** and click **Go**.
3. Under Tools click **Service Activation**.
4. Check if the Service “Cisco CallManager” is activated.

![Figure 3](http://www.yeastar.com)
3. VoIP Mode

If you choose VoIP mode, the FXS ports will be registered as the CUCM's SIP extensions, whose functions are the same with other SIP extensions on CUCM.

3.1. Create an extension(Directory number) on CUCM.

Step1: Create one extension number (called "Directory Number" on CUCM)

1) Go to Call Routing → Directory Number
2) Click Add New.
In the field "Directory Number" type in your extension number, leave everything else as it is.

![Directory Number Configuration](image)

3) Click Save.

Step2: Add an "End User".

1) Go to User Management → End User
2) Click Add New.

![End User Configuration](image)

3) Click Save.
Step3: Add the "End User" to the permission group.

1) Go down on the just saved End User page to "Permissions Information".
2) Click Add to The User Group. A pop-up window opens.
3) Click Find and choose "Standard CCM End Users" and click Add Selected.

Figure 6

4) Click Save.

Step4: Set TA3200 as a "Third Party SIP Device (Basic)" on CUCM.

1) Go to Device→Phone
2) Click **Add new**
3) Choose within the **Phone Type** list "Third Party SIP Device (Basic)"

4) Click **Next**
5) Check the MAC-address of TA3200. (It is written at a sticker on your device; for example: MAC: F4B549F0019C)
6) Type in only the numbers and letters in the field **MAC Address** (e.g. F4B549F0019C)
7) In **Phone Button Template** field select "Third-party SIP Device (Basic)".
8) In **Owner User ID** field select "111111", which is the directory number we created above.
9) In **Device Security profile** field select "Third-party SIP Device Basic-Standard Non-Secure Profile"
10) In **SIP Profile** field select the "Standard SIP Profile".
11) In **Digest User** field select "111111", which is the directory number we created above.

**Note:** Each device that is configured in Cisco Unified CM requires a unique digest user ID. When the device sends the REGISTER paket, Cisco Unified CM will immediately respond with a 401 challenge to get the Authentication header. The system uses the user ID from the authentication header to find the configuration entry in the database. If the third-party phone is not configured with the correct user ID, or the user ID is not associated with the device in the Cisco Unified CM database, Cisco Unified CM will respond with a 404 Not Found.
12) Click **Save**.
13) Click **Apply Config**.
Step 5: Associating your device with your extension and user.

1) In the left corner of the just saved page appears a canvas "Associate Information".
2) Click the link inside "Line [1] - Add a new DN".

3) In the field Directory Number type in your extension number (e.g. 111111).

4) The server recognizes the existing number and the data of your extension are going to appear.

5) At the end of the page click the button Associate End Users. A pop-up window appears.
6) Click **Find**.

![Figure 12](image)

7) Check your created user (e.g. 111111).
8) Click **Add Selected**.
9) Click **Close**.
10) At the End of the page you can check if your user is associated correctly.
11) Click **Save**.

3.2. **Register the extension(Directory number) on TA3200.**

**Step 1. Configure one VoIP server template on NeoGate TA3200.**

**Path:** Gateway→VoIP Settings→VoIP Server Settings
Server Name: CUCM  
Type: SIP  
Enable Register: Checked  
Transport: UDP  
Hostname/IP: fill in the CUCM IP address, 192.168.10.200  
Domain: fill in the CUCM IP address, 192.168.10.200

**Step 3. Edit the Dial Pattern Template**

**Path:** Gateway→VoIP Settings→VoIP Server Settings

The default dial pattern is set as ".", which allows you to dial any number out. In this guide, we will remain the default setting. You can change it according to your environment.

---

**Step 4. Edit the FXS port**

**Path:** Gateway→FXS Port List→FXS Port List
**Primary Server:** choose “CUCM(1)”  
**User Name:** the extension username on CUCM, 111111.  
**Authentication Name:** the extension authentication name on CUCM, 111111.  
**Password:** the authentication password of the extension 111111 on CUCM.  
**Dial Pattern Template:** choose the Dial Pattern Template: DialPatternTemplate1(1).

Save and apply the changes, and you will see the port status is “Registered” on "Port Status" page.

**Path:** Status → System Status → Port Status

Now, you are able to use the analog phone which is connected to NeoGate TA's FXS port 6 to make calls and receive calls.  
**Note:** ALL outgoing calls to CUCM extensions and to external numbers through trunks on CUCM should match the route pattern, or calls will fail.
4. SPS/SPX Mode

If you choose this mode to connect NeoGate TA3200 and CUCM, the FXS port will be registered as a Service Provider SIP trunk (SPS) to the CUCM. One SPS trunk to NeoGate TA also should be created on CUCM.

4.1 TA3200 Configuration

Step 1. Edit one VoIP Server template as SPS mode.

Path: Gateway→VoIP Settings→VoIP Server Settings

Do not check "Enable Register", choose SIP protocol, and fill in the CUCM IP address, the VoIP server template will be configured as SPS mode.

![Edit VoIP Server - VolPServer4](image)

**Figure 17**

- **Server Name**: CUCM
- **Type**: SIP
- **Enable Register**: DO NOT check
- **Transport**: UDP
- **Hostname/IP**: fill in CUCM IP address, 192.168.10.200
- **Domain**: fill in CUCM IP address, 192.168.10.200
Step 2. Edit the Dial Pattern Template

**Path:** Gateway → VoIP Settings → VoIP Server Settings

The default dial pattern is set as ".", which allows you to dial any number out. In this guide, we will remain the default setting. You can change it according to your environment.

![Edit Dial Pattern Template](image)

Figure 18

Step 3. Edit the FXS port

**Path:** Gateway → FXS Port List → FXS Port List

![Edit FXS Port](image)

Figure 19
**Number**: set a number for the FXS port. The number should be different from the extension numbers on CUCM. Here we set number 100 for FXS port 2.

**Primary Server**: choose CUCM(4), the VoIP server template configured on step 2.

**Dial Pattern Template**: choose the Dial Pattern Template, DialPatternTemplate1(1).

**Note**: You don't need to fill in any authentication name and authentication password on the FXS port edit page if you choose SPS/SPX mode.

After saving and applying the changes, you will see the trunk is “OK” in “Line Status”.

**Path**: Status→System Status→Port Status

![Figure 20](image-url)

### 4.2 CUCM Configuration

**Step 1**: Configure the SIP Trunk Security Profile

1) Go to **System→Security→SIP Trunk Security Profile**

2) Click **Add New**.

3) Configure the fields as follows:

   Please change the “Outgoing Transport Type” to UDP, otherwise the outgoing calls would fail.
In version 9.x, the Non Secure SIP Trunk Profile already exists, but it must be modified.

1) On Unified CM, go to **System** → **Security** → **SIP Trunk Security Profile**.
2) Select **Non Secure SIP Trunk Profile**.
3) Modify the fields as follows:

4) Click **Save**.
4) Click **Save**.

**Step2:** Configure the SIP Profile

1) On Unified CM, go to **Device**→**Device Settings**→**SIP Profile**.
2) Click **Copy** against the Standard SIP Profile.
3) Check “Allow Presentation Sharing using BFCP” if BFCP (Dual video / presentation sharing) is required. Check “Use Fully Qualified Domain in SIP Requests” if needed, leave everything else as it is:
## SIP Profile Configuration

### Status
- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take effect.

### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Standard SIP Profile for TA3200</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>Default SIP Profile</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>None</td>
</tr>
<tr>
<td>Early Offer for G-Clear Calls</td>
<td>Disabled</td>
</tr>
<tr>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites</td>
<td>TIAG and AS</td>
</tr>
<tr>
<td>User-Agent and Server header information</td>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Redirect by Application</td>
<td></td>
</tr>
<tr>
<td>Disable Early Media on 100</td>
<td></td>
</tr>
<tr>
<td>Outgoing T.38 INVITE include audio rline</td>
<td></td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Require SDP Inactive Exchange for Mid-Call Media Change</td>
<td></td>
</tr>
<tr>
<td>Use Fully Qualified Domain Name in SIP Requests</td>
<td></td>
</tr>
</tbody>
</table>

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>160</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>5600</td>
</tr>
<tr>
<td>Timer T1 (msce)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msce)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-oisco-serviceur-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Offer URI</td>
<td>x-oisco-serviceur-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-oisco-serviceur-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-oisco-serviceur-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Normal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>110</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>6</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To Find Digit Timer (milliseconds)</td>
<td>19000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-oisco-serviceur-callfw</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-oisco-serviceur-abrdial</td>
</tr>
</tbody>
</table>
Step 3: Configure the SIP Trunk device

1) On Unified CM, go to **Device** → **Trunk**.
2) Click **Add New**.
3) Select a Trunk Type of SIP Trunk.

4) Click **Next**.
5) Configure the Device Information fields as follows:
   
   **Mark the field modified compared with default settings.**
Trunk Configuration

### Device Information

- **Product:** SIP Trunk
- **Device Protocol:** SIP
- **Trunk Service Type:** http
- **Device Name:** Fan_02321
- **Description:**
- **Common Device Configuration:**
- **Call Classification:**

### Media Resource Group List

- **Location:**
- **AAR Group:**
- **Transport Protocol:**
- **QoS Version:**
- **ARQ-RRO TYPE:**
- **Packet Capture Mode:**
- **Packet Capture Duration:**

### Intercompany Media Engine (IME)

- **CUE Transformation Profile:**

### Interception Precedence and Prescription (IPPI) Information

- **RUDP Domain:**

### Call Routing Information

- **Incoming Profile Id:**
- **Contact Identity:**
- **Assigned Trunk:**
- **GTP Profile:**

### Dialed Numbers

- **Significant Digit:**
- **Preset Dial Tone:**
- **InDialed:**
- **OutDialed:**
- **Plan:**

### Redirecting Diameter Header (Deliver-302)

If the administrator sets the prefix to 302 this indicates that processing will use prefix at the next level setting (Service/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

### Connected Profile Settings

- **Connected Party Linking:**
- **Use Service Port Connected Profile:**
- **Use Service Port Connected Service Profile:**

For more information, please visit [http://www.yeastar.com](http://www.yeastar.com).
6) Click **Save**.
7) Click **Reset**.

**Step 4:** Configure Route Pattern for the SIP Trunk.
1) On Unified CM, go to **Call Routing** → **Route/Hunt** → **Route Pattern**.
2) Click **Add New**.
3) Configure Route Pattern as follows:
   
   Set Route Pattern to 1XX, which means all 3-digit calls starting with 1 will be sent to FXS gateway via this SIP Trunk. Besides, you can set up multiple Route Patterns for a SIP trunk.
Figure 26

4) Click **Save**.

**Step5:** Check the status of this trunk.
On Unified CM, go to **Device→Trunk**, click **Find**.

In this guide, you should dial 100 on CUCM to call the phone which is connected to TA3200 FXS port 2.

<End>