

Grandstream Networks, Inc.

SIP Trunks Guide

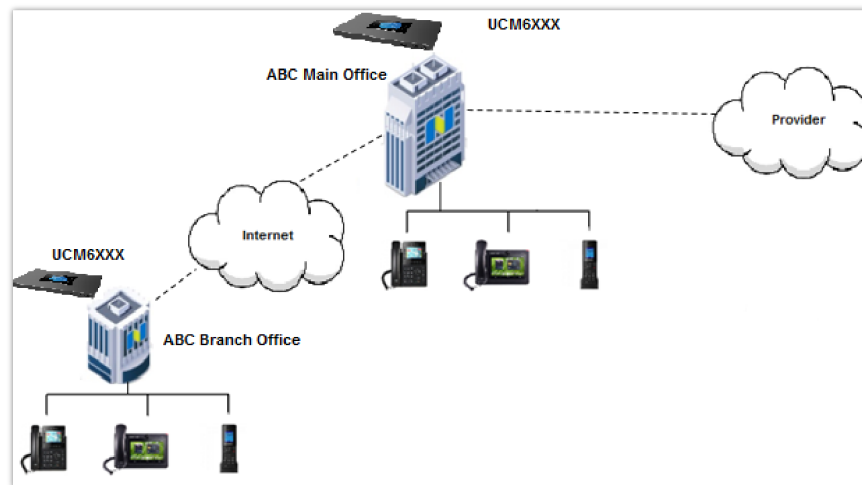


SIP trunks are a VoIP service that can be provided from an ITSP (Internet Telephony Service Provider) to extend telephony features beyond IPPBX local area. SIP trunks can carry voice calls, video calls, instant messages, multimedia conferences, and other SIP-based, real-time communications services.

Using SIP trunks helps to reduce call rates especially when making long distance calls, since VoIP providers can offer better calling rates compared to local ISP using analog lines.

UCM6xxx series support two types of SIP trunks: "Register SIP trunks", mainly used to connect with provider's trunk and "Peer trunks", that can be used to interconnect multiple IP-PBXs. UCM6xxx series support up to 200 SIP trunks.

This guide describes needed configuration to set up register trunk (with provider) and peer trunk (between two UCM6xxx).



UCM6xxx Typical Scenario

The figure above shows a typical scenario using two UCM6xxx between main and branch offices (connected via peer trunk), and main office UCM6xxx connected to provider via register SIP trunk.

UCM6XXX series include UCM620x series, UCM630x series, UCM630xA series and UCM6510.

Account SIP Trunks

Account SIP Trunks allow creating a SIP trunk which allow devices like ATAs and gateways without a static IP addresses to connect to the UCM to handle and route calls. This type of SIP trunks supports authentication which requires providing the correct username and password to be able to register on the trunk. This trunk can be used for scenarios where devices either do not have a static IP address to peer with, or cannot register to a UCM extension due to call handling and routing complications.

Configuration

To create an Account SIP Trunk, please navigate to **Extensions/Trunk > VoIP Trunk**, then click on "Add".

VoIP Trunks > Edit SIP Trunk: Grandstream

Basic Settings

Advanced Settings

If the host is not a numeric IP address, but the port number is present in the URI, the UCM performs an A or AAAA record lookup of the domain name. If a domain is configured without a port number, the UCM will do an SRV record lookup.

Disable This Trunk ☐

Provider Name

Transport

ⓘ If TCP or TLS is selected, please enable the corresponding settings in [PBX Settings -> SIP Settings -> TCP/TLS](#).

Auto Record ☐

Keep Original CID ☐

Keep Trunk CID ☐

NAT ☒

TEL URI

CallerID Number

Cancel

Save

Account SIP Trunk

The user must enter the name of the provider, then enter the username and password for authentication. Both are mandatory to be able to create this type of trunk.

In order to register a PBX to this SIP trunk, please create a SIP trunk that supports registration. In the case of the UCM6300 series, we will create a Register Trunk which support authentication. Please follow the steps below to configure the Register Trunk properly.

1. On the Register Trunk, enter the Host Name of the PBX on which we have created the Account SIP Trunk. The user can enter the PBX's domain name (if applicable) or its IP address as shown in the figure below.

* Provider Name

* Host Name

Transport

ⓘ If TCP or TLS is selected, please enable the corresponding settings in [PBX Settings -> SIP Settings -> TCP/TLS](#).

Host Name Configuration

2. Enter the username and password which are the configured on the Account SIP Trunk.

* Username

* Password

Authentication Information Configuration

To configure the inbound and outbound routes, please refer to the following guide:
<https://documentation.grandstream.com/knowledge-base/how-to-manage-inbound-outbound-routes/>

Register SIP Trunks

In this guide, we will take company ABC as example. The company has purchased two SIP trunks, the first one will be used for International calls, while the second will be used of national ones.

The ITSP provider gave following trunk information to the company ABC in order to register their trunks.

SIP Trunk Information	Value
SIP Trunk 1	
Provider address	sip.provider.com
Username	0655441000
Authenticate ID	0655441000
Password	admin123
Main number	0655441000
Provided DID's	0655441001 / 0655441002 / 0655441003 / 0655441004 / 0655441005 / 0655441006 / 0655441007 / 0655441008 / 0655441009 / 0655441010
SIP Trunk 2	
Provider address	sip.provider.com
Username	0655441100
Authenticate ID	0655441100
Password	admin123
Main number	0655441100
Provided DID's	0655441101 / 0655441102 / 0655441103 / 0655441104 / 0655441105 / 0655441106 / 0655441107 / 0655441108 / 0655441109 / 0655441110

Setup considerations:

- Extensions range on UCM6xxx in main office is 1000 – 1999.
- Dialing to international numbers should be prefixed with 99 with no size limits.
- National numbers start with 06 with 10-digit length.

Configuration

Below steps to configure SIP Trunk 1 on Main office UCM6xxx. Same steps apply to configure Trunk 2.

1. Access UCM6xxx web GUI → **Extension / Trunk** → **VoIP Trunks**.
2. Click on **+ Add SIP Trunk** , in the following screenshot type in Trunk 1 credentials:

VoIP Trunks > Create New SIP Trunk

If the host is not a numeric IP address, but the port number is present in the URI, the UCM performs an A or AAAA record lookup of the domain name. If a domain is configured without a port number, the UCM will do an SRV record lookup.

Disable This Trunk ☐

Type **Register SIP Trunk**

* Provider Name **Please select a provider**

* Host Name

Transport **UDP**

ⓘ If TCP or TLS is selected, please enable the corresponding settings in PBX Settings -> SIP Settings -> TCP/TLS.

Keep Original CID ☐

Keep Trunk CID ☒

NAT ☐

TEL URI **Disabled**

Need Registration ☒

Allow outgoing calls if registration fails ☒

CallerID Name

Cancel **Save**

Create New Register Trunk

- Select **"Register SIP Trunk"** as Type.
- Type in a reference name for **Provider Name**. For example: "Provider_1".
- Enter provider's FQDN/IP address in **Host Name** field. In this example: "sip.provider.com".
- Enter **Username** to register to the provider. In this example: "0655441000".
- Enter **Authenticate ID** to register to the provider. In this example: "0655441000".
- Enter **Password** associated with username. In this example: "admin123".

3. Click **Save** and **Apply Changes** to store and apply the configuration.

Above steps describe basic configuration needed to register a SIP trunk. Depending on providers, users may need to adjust their settings to successfully register a SIP trunk.

The table below describes main parameters available for Register SIP Trunks:

Basic Settings	
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "UDP". • UDP • TCP

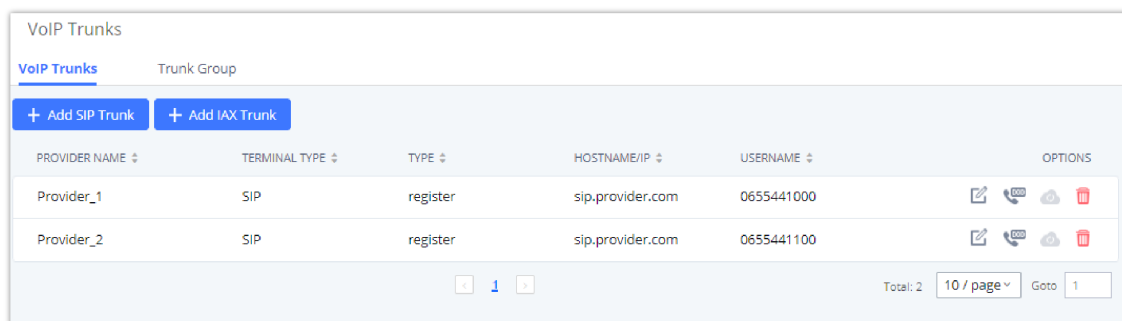
	• TLS
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded. Note:
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override the “Keep Trunk CID” option. Please make sure that the peer PBX at the other side supports to match user entry using the “username” field from the authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension’s CID when the extension has CID configured. The default setting is “No”.
NAT	Turn on this option when the PBX is using public IP and communicating with devices behind NAT. If there is a one-way audio issue, usually it is related to NAT configuration or SIP/RTP port configuration on the firewall.
Disable This Trunk	If selected, the trunk will be disabled. Note: If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled.
Need Registration	Select whether the trunk needs to register on the external server or not when the "Register SIP Trunk" type is selected. The default setting is No.
Allow outgoing calls if registration failure	If enabled outgoing calls even if the registration to this trunk fails will still be able to go through. Note that if we uncheck the “Need Registration” option, this option will be ignored.
CallerID Name	Configure the new name of the caller when the extension has no CallerID Name configured.
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the “From” Header. For example, "trunk.UCM630X.provider.com" is the From Domain in From Header: sip:1234567@trunk.UCM630X.provider.com.
From User	Configure the actual username of the extension. This can be used to override the “From” Header. There are cases where there is a single ID for registration (single trunk) with multiple DIDs. For example, "1234567" is the From User in From Header: sip:1234567@trunk.UCM630X.provider.com.
Username	Enter the username to register to the trunk from the provider when the "Register SIP Trunk" type is selected.
Password	Enter the password to register to the trunk when "Register SIP Trunk" is selected.
Auth ID	Enter the Authentication ID for the "Register SIP Trunk" type.
Auth Trunk	If enabled, the UCM will send a 401 response to the incoming call to authenticate the trunk.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under Web GUI→CDR→Recording Files.
RemoteConnect Mode	If enabled, the RemoteConnect related parameters will be set synchronously. Please make the trunk host is allocated by GDMS or it supports TLS.

Direct Callback	Allows external numbers the option to get directed to the extension that last called them. For Example, User 2002 has dialed external number 061234575, but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option “Direct Callback” is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.
Enable Concurrent Call Alert	If enabled, the "Trunk Concurrent Calls" system event will monitor the number of concurrent calls in this trunk. When the number of concurrent calls in a certain period exceeds the set threshold, an alarm message will be generated. Note: Please turn on the alert for the "Trunk Concurrent Calls" event first.
Two-way Concurrent Calls Threshold	Threshold of all incoming and outgoing concurrent calls through this trunk.
Outgoing Concurrent Calls Threshold	Threshold of all outgoing concurrent calls passing through this trunk.
Incoming Concurrent Calls Threshold	Threshold of all incoming concurrent calls passing through this trunk.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Send PPI Header	If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is “No”. Note: “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers can be contained in the SIP INVITE message.
PPI Mode	● Default: Include the trunk’s preferred CID (configured in Basic Settings) in the PPI Header. ● Original CID: Include the original CID in the PPI Header. ● DOD Number: Include the trunk’s DOD number in the PPI Header. If no DOD number has been set, the trunk’s preferred CID will be used.
Send PAI Header	If enabled, the SIP INVITE message sent to the trunk will contain PAI (P-Asserted-Identity) header including configured PAI Header. The default setting is “No”. Note: “Send PPI Header” and “Send PAI Header” cannot be enabled at the same time. Only one of the two headers can be contained in the SIP INVITE message.
PAI Header	If “Send PAI Header” is enabled and “PAI Header” is configured as “123456” for instance, the PAI header in the SIP message sent from the UCM will contain “123456”. If “Send PAI Header” is enabled and “PAI Header” is configured as “empty”, the PAI header in the SIP message sent from the UCM will contain the original CID. Note: “Send PAI Header” needs to be enabled to use this feature
Send Anonymous	If checked, the "From" header in the outgoing INVITE message will be set to anonymous.
DOD As From Name	If enabled and "From User" is configured, the INVITE's From header will contain the DOD number. Note:
Passthrough PAI Header	If checked and the option "Send PAI Header" is not checked, the PAI header will be passthrough from one side to the other side.

Send PANI Header	If checked, the INVITE and REGISTER sent to the trunk will contain the P-Access-Network-Info header.
Access Network Info	The access network information is in the P-Access-Network-Info header.
Send Anonymous	If checked, the "From" header in the outgoing INVITE message will be set to anonymous.
Outbound Proxy Support	Select to enable outbound proxy in this trunk. The default setting is "No".
Outbound Proxy	When outbound proxy support is enabled, enter the IP address or URL of the outbound proxy.
Remove OBP from Route	It is used to set if the phone system will remove outbound proxy URI from the route header. If is set to “Yes”, it will remove the route header from SIP requests. The default setting is “No”.
DID Mode	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line".
GIN Registration	If enabled, the UCM will send a GIN REGISTER (generate implicit numbers).
DTMF Mode	Configure the default DTMF mode when sending DTMF on this trunk. ● Default: The global setting of DTMF mode will be used. The global setting for the DTMF Mode setting is under Web GUI→PBX Settings→SIP Settings→ToS. ● RFC4733: Send DTMF using RFC4733. ● Info: Send DTMF using SIP INFO message. ● Inband: Send DTMF using inband audio. This requires a 64-bit codec, i.e., PCMU and PCMA. ● Auto: Send DTMF using RFC4733 if offered. Otherwise, an inband will be used.
Enable Heartbeat Detection	If enabled, the UCM630X will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Heartbeat Frequency	When the "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
The Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default setting is 0, which means no limit.
Packet Loss Retransmission	Configure to enable Packet Loss Retransmission.
Audio FEC	Configure to enable Forward Error Correction (FEC) for audio.
Video FEC	Configure to enable Forward Error Correction (FEC) for video.
ICE support	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.
TURN Relay	Enable TURN Relay This option will allow using TURN servers for media NAT traversal. This option will also be prioritized if ICE Support is already enabled.
FECC	Configure to enable Far-end Camera Control
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
SRTP Crypto Suite	The following encryption protocols can be used to encrypt an RTP stream.

	• AES_CM_128_HMAC_SHA1_80 (This is the default used protocol) • AES_256_CM_HMAC_SHA1_80 • AEAD_AES_128_GCM • AEAD_AES_256_GCM
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.
STIR/SHAKEN	Configure this feature to prevent call spoofing, robocalls and spam calls. • Disabled • Outgoing Attest: The UCM will only authenticate the caller ID. • Incoming Verify: The UCM will only verify if the caller ID is authenticated • Both: The UCM will authenticate the outgoing calls as well as verify if the incoming calls are authenticated.
CC Settings	
Enable CC	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason.
CC Max Agents	Configure the maximum number of CCSS agents which may be allocated for this channel. In other words, this number serves as the maximum number of CC requests this channel is allowed to make. The minimum value is 1.
CC Max Monitors	Configure the maximum number of monitor structures that may be created for this device. In other words, this number tells how many callers may request CC services for a specific device at one time. The minimum value is 1.

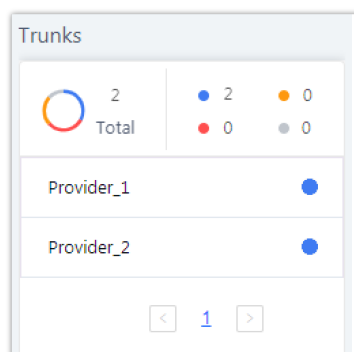
After saving the new VoIP trunk, it will be displayed under **Web UI → Extension / Trunk → VoIP Trunks** as shown below. Press  to edit trunk settings if needed.



VoIP Trunks					
VoIP Trunks Trunk Group					
+ Add SIP Trunk		+ Add IAX Trunk			
PROVIDER NAME	TERMINAL TYPE	TYPE	HOSTNAME/IP	USERNAME	OPTIONS
Provider_1	SIP	register	sip.provider.com	0655441000	   
Provider_2	SIP	register	sip.provider.com	0655441100	   
< 1 > Total: 2 10 / page Goto: 1					

Register Trunk

Once the trunk has been created, users can check its registration status under **System Status → Dashboard** page.



Trunks	
 2 Total	2 0 0 0
Provider_1	
Provider_2	
< 1 >	

Register Trunk Status

Note

If status shows "Rejected", it means that UCM6xxx didn't get registered with the provider. Verify that provider's server is reachable from UCM6xxx and double confirm trunk credentials.

DID/DOD Configuration

Direct Inward Dialing (DID)

DID (Direct Inward Dialing) is a service provided by the ITSP to subscribers using IP-PBX, adding possibility to route incoming calls to a specific DID to a specific extension. Using DIDs, it will allow extensions to be reachable from outside directly without going through main office number.

In this example, following DIDs have been provided with first trunk "Provider_1": 0655441000 / 0655441001 / 0655441002 / 0655441003 / 0655441004 / 0655441005 / 0655441006 / 0655441007 / 0655441008 / 0655441009 / 0655441010.

Company wants to redirect incoming calls to specific extensions using provided DIDs.

UCM6xxx in main office is using extensions range: 1000 – 1999.

Note

Make sure **DID Mode** is set correctly under trunk advanced settings. Provider may include DID number in "Request-line" or "To-header".

Following configuration can be done to achieve this:

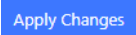
1. Access UCM6xxx Web UI → **Extension/Trunk** → **Inbound Routes**. Select "Trunk 1" and press **"Add"** button.
2. In **DID Pattern** field, enter "_0655441xxx".
3. Configure **ByDID** as **Default Destination** and check "Extension" in "DID Destination" options.
4. Configure **Strip** field. In this example: Strip=6 to remove first 6 digits from incoming number and keep last 4 digits. If call is coming to 0655441007, UCM6xxx will strip 065544 and keep 1007.
5. Click **Save** and **Apply Changes** to apply configuration.

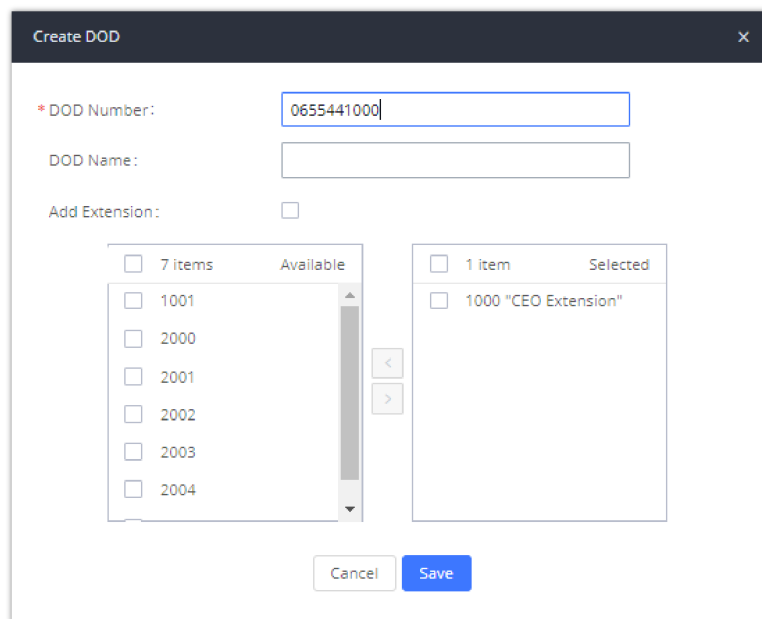
Direct Outward Dialing (DOD)

The UCM6xxx provides Direct Outward Dialing (DOD) which is a service of a local phone that allows extensions within a company ABC's UCM to connect to outside lines directly.

We will use Company's ABC trunk 0655441000 with 11 DID's associated to it. At the moment when a user makes an outbound call their caller ID shows up as the main office number which is 0655441000. This create a problem as the CEO would like that his calls to comes from his direct line. This can be accomplished by configuring DOD for the CEO's extension. Other group member also can benefit from DOD to have their own line showed when making calls.

Steps on how to configure DOD on the UCM6xxx:

1. To setup DOD go to UCM6xxx Web UI → **Extension/Trunk** → **VoIP Trunks** page.
2. Click  to access the DOD options for the selected SIP Trunk.
3. Click **" + Add DOD "** to begin your DOD setup.
4. For **"DOD Number"**, enter one of the numbers (DIDs) from your SIP trunk provider. In this example, we will enter in the number for the CEO's direct line (0655441000).
5. Enter a name for the DOD in the **"DOD Name"**.
6. If extension number need to be appended to the DID number click on **"Add Extension"**.
7. Select an extension from the **"Available Extensions"** list. Users have the option of selecting more than one extension. In this case, we would select the CEO's extension. After making the selection, click on the  button to move the extension(s) to the **"Selected Extensions"** list.
8. Click  and  to apply configuration.

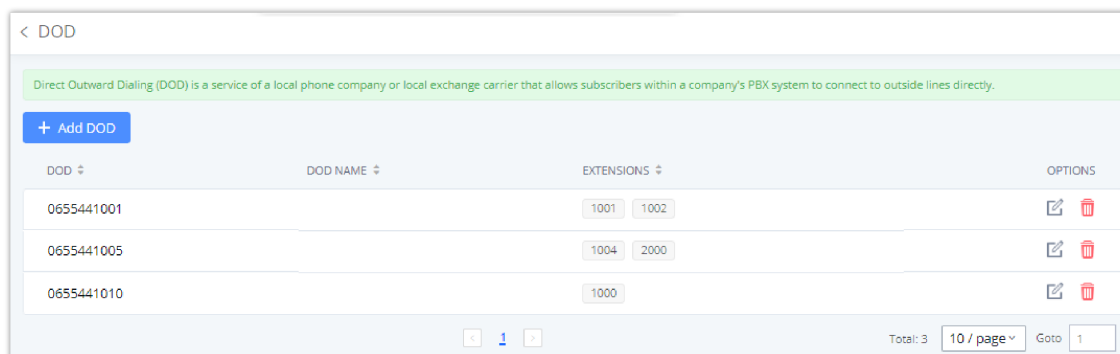


The 'Create DOD' dialog box contains the following fields and controls:

- * DOD Number:** Text input field containing '0655441000'.
- DOD Name:** Text input field.
- Add Extension:** A checkbox.
- Available Extensions:** A list box showing 7 items: 1001, 2000, 2001, 2002, 2003, 2004. Each item has a checkbox.
- Selected Extensions:** A list box showing 1 item: 1000 "CEO Extension". It has a checkbox.
- Navigation:** Left and right arrow buttons between the two lists.
- Buttons:** 'Cancel' and 'Save' buttons at the bottom.

Create New DOD

Users can press **"Edit DOD"** to check/add/delete extensions that are associated to a particular DOD.



The 'DOD List' table displays the following data:

DOD	DOD NAME	EXTENSIONS	OPTIONS
0655441001		1001 1002	 
0655441005		1004 2000	 
0655441010		1000	 

Page navigation: Total: 3, 10 / page, Goto 1

DOD List

Create Outbound/Inbound Routes

In this section we will give an example of creating Outbound and Inbound Rules for trunk 1 "Provider_1" to allow receiving incoming calls and to be redirected to IVR and to allow outgoing calls to be made.

Create Inbound Routes

1. Access UCM6xxx Web UI → **Extension/Trunk** → **Inbound Routes**. Select "Trunk 1" and press "+ Add".
2. In **DID Pattern** field, enter "_X.". This allows all incoming calls to be received.
3. Set **IVR** as **Default Destination**. (Assuming IVR is pre-configured).
4. Click **Save** and **Apply Changes** to apply configuration.

Create New Inbound Rule

Trunks: SIP Trunks -- 123Cloud

Pattern: _X.

Inbound Route Name:

CallerID Pattern:

CID Source: None

Alert-Info: None

Ringback Tone: None

Auto Record:

Prepend Trunk Name:

Enable Route-Level Inbound Mode:

Disable This Route:

Seamless Transfer Whitelist:

Fax Detection:

Block Collect Calls:

Set CallerID Info:

Inbound Multiple Mode:

Default Mode Mode 1

Default Destination: IVR Main

Time Condition

Add

TIME CONDITION	TIME	WEEK	MONTH	DAY	DESTINATION	OPTIONS
----------------	------	------	-------	-----	-------------	---------

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

Register Inbound Route

Note

On UCM630x series, you can also configure the name of the Inbound Route Name, it is optional.

Edit Inbound Rule

Inbound Route Name: Inbound_route_name

Pattern: _X.

Disable This Route:

Inbound Route name Field on UCM630x

- As for time condition based inbound route, please refer to section **TIME CONDITION**.
- **Special ringing tone** option is called **Ringback tone** on the UCM63xx series.

Create Outbound Routes

1. Go to **Web UI → Extension/Trunk → Outbound Routes**.
2. Click on **+ Add** button to create a new outbound route.
3. Enter the **Calling Rule Name, Pattern** and choose **"Provider_1"** in **Use Trunk**.

In below figure, we set "Pattern" to "_06XXXXXXXX" to allow only dialed numbers with leading 06 and 10-digit length to be accepted. Extensions with privilege "Internal" or higher can use this route/trunk.

Users can add comments to a dial plan by typing **"/**"** and **"*/"** before and after each comment respectively, in our example we can set the pattern to: **"_06XXXXXXXX /*route1*/"**

The screenshot shows the 'Create New Outbound Rule' window with the following configuration:

- General**
 - Outbound Rule Name: Provider1
 - Pattern: _06XXXXXXXX
 - PIN Groups: None
 - Password: (empty)
 - Local Country Code: (empty)
 - Disable This Route: ☐
 - Privilege Level: Internal (Warning: Setting privilege level as "Internal" has potential security risks.)
 - PIN Groups with Privilege Level: ☐
 - Auto Record: ☐
- Enable Source Caller ID Whitelist**
 - Enable Source Caller ID Whitelist: ☐
 - Outbound Route CID: (empty)
- Call Duration Limit**
 - Call Duration Limit: ☐
- Main Trunk**
 - Trunk: SIP Trunks -- 123Cloud
 - Strip: (empty)
 - Prepend: (empty)

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

Outbound Route Example

Another outbound route should be set using "Provider_2" with following parameters:

- **Pattern:** "_99X." (to allow only dialed numbers with leading 99 to be authorized).
- **Privilege:** International for instance.
- **Strip:** "2" (to remove prefix 99 before calling out).

To dial international numbers such as 0016175669300, users need to have privilege "International" in used extension and dial 990016175669300.

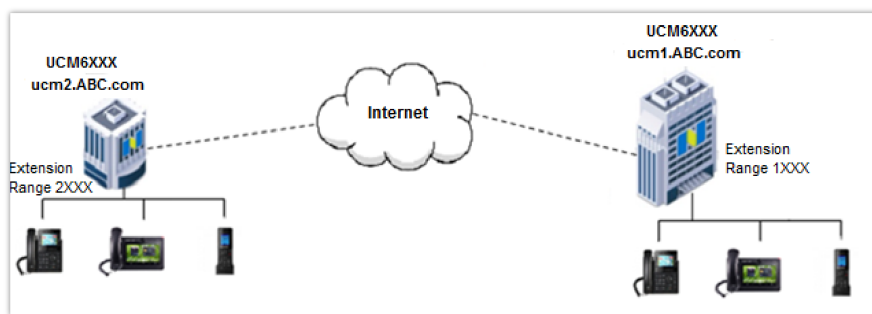
For time condition based Outbound Routes, please refer to TIME CONDITION.

Info

When DOD Number and Name configured, they are sent in the From and Contact headers in the SIP messaging.

Peer Trunk

In this chapter, we will consider that company ABC has a branch office equipped with UCM6xxx, and they want to be able to communicate between the two offices.



SIP Peer Trunk

Assuming following:

- Both UCM6xxxs have public IP addresses.
- Or both UCM6xxx are on the same LAN/VPN using private or public IP addresses.
- Or both UCM6xxx can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).
- UCM6xxx in main office is using extensions range 1XXX, while UCM6xxx in branch office is using extensions range 2XXX.

Configuration

Following steps need to be done on both UCM6xxx in both locations:

1. Access UCM6xxx's web GUI → **Extension / Trunk** → **VoIP Trunks**.
2. Click on **+ Add SIP Trunk**, and enter following parameters:
 - Select "Peer SIP Trunk" for Type.
 - Enter a reference name. In this example: **toBranch**.
 - In **Host Name** field, enter the IP address/domain of the other UCM6xxx. In this case "ucm2.abd.com" is the domain name of the branch office.
3. Click **Save** and **Apply Changes** to apply configuration.

Create New SIP Trunk

CancelSave

If the host is not a numeric IP address, but the port number is present in the URI, the UCM performs an A or AAAA record lookup of the domain name. If a domain is configured without a port number, the UCM will do an SRV record lookup.

Type:Peer SIP Trunk

Provider Name:To_Branch

Host Name:ucm2.abc.com

Transport:UDP

Keep Original CID:☐

Keep Trunk CID:☐

NAT:☐

Disable This Trunk:☐

TEL URI:Disabled

CallerID Number:

CallerID Name:

Auto Record:☐

Direct Callback:☐

RemoteConnect Mode:☐

Monitor Concurrent Calls:☐

Concurrent Call:

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

Create New Peer Trunk

The tables below describe basic and advanced parameters available for PEER trunks:

Field	Description
Provider Name	Configure a unique label to identify the trunk when listed in outbound rules and incoming rules.
Host Name	Enter the IP address or hostname of the VoIP provider's server.
Transport	Configure the SIP Transport method. Using TCP requires local TCP support; using TLS requires local TLS support.
Auto Record	Enable automatic recording for the calls using this trunk (for SIP trunk only). The default setting is disabled. The recording files can be accessed under web GUI->CDR->Recording Files
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using "username" field from authentication line. If enabled, user at extension 1000 will see 2000 as UserID.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No". If enabled and Keep Original CID is disabled, user at extension 1000 will see the ID set on Caller ID field.
NAT	Turn on this setting when the PBX is using public IP and communicating with devices behind NAT. If there is one-way audio issue, usually it is related to NAT configuration or SIP/RTP port support on the firewall.

Field	Description
Disable This Trunk	If checked, the trunk will be disabled. Note: If a current SIP trunk is disabled, UCM will send UNREGISTER message (REGISTER message with expires=0) to the SIP provider.
TEL URI	If the trunk has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and TO header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is disabled. ◦ If TEL URI is set to Disabled, the TO header will be "To: <sip:1000@ucm1.abc.com>". ◦ If TEL URI is set to User=Phone, the TO header will be "To: <sip:1000@ucm1.abc.com;user=phone>". ◦ If TEL URI is set to Enabled, the TO header will be "To: <tel:1000>".
Caller ID Number	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. Important Note: When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call : ◦ CID from inbound call (Keep Original CID Enabled) → Trunk Username/CallerID (Keep Trunk CID Enabled) → DOD → Extension CallerID Number → Trunk Username/CallerID (Keep Trunk CID Disabled) → Global Outbound CID.
Caller ID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured. Note: This Option needs Keep Trunk CID Feature enabled ◦ If the Caller ID Name is set on the trunk, the "From" header will include this name as shown below: From: "GSTest" <sip:2000@ ucm2.abc.com >;tag=f268
From Domain	Configure the actual domain where the extension comes from.
Jitter Buffer (UCM630x/A only)	Select jitter buffer method for temporary accounts such as meeting participants who joined via link. ◦ Disable: Jitter buffer will not be used. ◦ Fixed: Jitter buffer with a fixed size (equal to the value of "Jitter Buffer Size") ◦ Adaptive: Jitter buffer with an adaptive size that will not exceed the value of "Max Jitter Buffer"). ◦ NetEQ: Dynamic jitter buffer via NetEQ.
Direct Callback	Allows external numbers the option to get directed to the extension that last called them.
RemoteConnect Mode (UCM630x/A only)	If enabled, RemoteConnect-related options will be automatically configured. Please confirm the trunk is a GDMS-assigned address or supports TLS.
Enable Concurrent Call Alert (UCM630x/A only)	If enabled and when the number of concurrent calls exceeds any trunk's configured concurrent call thresholds, an alarm notification will be generated. Note: Please make sure the system alert event "Trunk Concurrent Calls" is enabled.

Field	Description
Concurrent Call Threshold (UCM630x/A only)	Threshold of all incoming and outgoing concurrent calls through this trunk. Please enter a value between 1 and 200.
Outgoing Concurrent Call Threshold (UCM630x/A only)	Threshold of all outgoing concurrent calls passing through this trunk. Please enter a value between 1 and 200.
Incoming Concurrent Call Threshold (UCM630x/A only)	Threshold of all incoming concurrent calls passing through this trunk. Please enter a value between 1 and 200.

Extension/Trunk → VoIP Trunks

Field	Description
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p. The selected codecs will be sent on the SDP field from top to bottom to the other UCM in order to negotiate the codec to use during the call.
Send PPI Header	If enabled, the SIP INVITE message sent to the trunk will contain PPI (P-Preferred-Identity) header. The default setting is "No". Note: "Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message. If enabled the SIP header will contain the following: P-Preferred-Identity: sip:2000@ ucm2.abc.com This feature is needed by the provider "some providers use PPI others use PAI"
Send PAI Header	If enabled, the SIP INVITE message sent to the trunk will contain PAI (PAsserted-Identity) header. The default setting is "No". Note: "Send PPI Header" and "Send PAI Header" cannot be enabled at the same time. Only one of the two headers is allowed to be contained in the SIP INVITE message. If enabled the SIP header will contain the following: P-Asserted-Identity: sip:2000@ucm2.abc.com This feature is needed by the provider "some providers use PPI others use PAI"
Passthrough PAI Header	If enabled and "Send PAI Header" is disabled, PAI headers will be preserved as calls pass through the UCM.
Send PANI Header (UCM630X/A only)	If checked, the INVITE sent to the trunk will contain P-Access-Network-Info header.
Send Anonymous (UCM630x/A only)	If checked, the "From" header in outgoing INVITE message will be set to anonymous.

Field	Description
DID Mode	Configure where to get the destination ID of an incoming SIP call, from SIP Request-line or To-header. The default is set to "Request-line". ◦ If set to Request-line, the UCM will extract the ID from the Request-Line of the incoming INVITE and set it on the "To header" for the outgoing one. ◦ If set to To-Header, the UCM will extract the ID from the To-Header of the incoming INVITE and set it on the "To header" for the outgoing one.
DTMF Mode	Configure the default DTMF mode when sending DTMF on this trunk. ◦ Default: The global setting of DTMF mode will be used. The global setting for DTMF Mode setting is under web UI->PBX->SIP Settings- >ToS. ◦ RFC2833: Send DTMF using RFC2833. ◦ Info: Send DTMF using SIP INFO message. ◦ Inband: Send DTMF using inband audio. This requires 64-bit codec, i.e., PCMU and PCMA. ◦ Auto: Send DTMF using RFC2833 if offered. Otherwise, inband will be used.
Enable Heartbeat Detection	If enabled the UCM will send regularly SIP OPTIONS to check if the device is online.
Heartbeat Frequency	When "Enable Heartbeat Detection" option is set to "Yes", configure the interval (in seconds) of the SIP OPTIONS message sent to the device to check if the device is still online. The default setting is 60 seconds.
Maximum Number of Call Lines	The maximum number of concurrent calls using the trunk. The default settings 0, which means no limit.
Packet Loss Retransmission (UCM630x/A only)	Configure to enable Packet Loss Retransmission.
Audio FEC (UCM630x/A only)	Configure to enable Forward Error Correction (FEC).
Video FEC (UCM630x/A only)	Enables Video FEC.
ICE Support (UCM630x only)	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.
FECC (UCM630x/A only)	Enable for Remote Camera Management.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No". ◦ If enabled it will provide encryption, message authentication and integrity for the Audio stream.
IPVT Mode	Configures the UCM to be used exclusively for IPVT. Warning: This will lock out certain UCM features.

Field	Description
Sync LDAP Enable	If enabled, the local UCM6xxx will automatically provide and update the local LDAP contacts to the remote UCM6xxx SIP peer trunk. In order to ensure successful synchronization, the remote UCM6xxx peer also needs to enable this option on the SIP peer trunk. The default setting is "No". ○ Password and Ports need to be the same on both UCM's in order to synchronize LDAP data. Please refer to UCM6xxx User Manual for more detail about this feature.
Enable T.38 UDPTL (UCM630x/A only)	Enables or disables T.38 UDPTL support.
STIR/SHAKEN (UCM630x/A only)	Blocks Spam Calls.
Enable CC	If enabled, the system will automatically alert the user when a called party is available, given that a previous call to that party failed for some reason. ○ For more information about this feature please refer to the https://documentation.grandstream.com/knowledge-base/busy-camp-on-2/

Extension/Trunk → VoIP Trunks → Advanced Settings

After clicking Save, the new VoIP trunk will be displayed under “**Web UI → Extension / Trunk > VoIP Trunks**” as shown below.

VoIP Trunks					
VoIP Trunks		Trunk Group			
+ Add SIP Trunk		+ Add IAX Trunk			
PROVIDER NAME	TERMINAL TYPE	TYPE	HOSTNAME/IP	USERNAME	OPTIONS
To_Branch	SIP	peer	ucm2.abc.com		 
		1		Total: 1 10 / page Goto 1	

Edit Peer Trunk

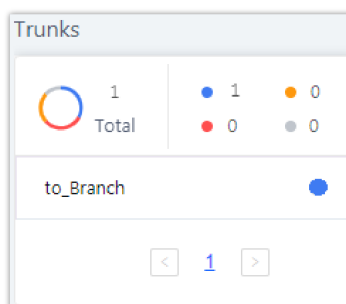
Press  and go to “Advanced Settings”. Check “**Enable Heartbeat Detection**” option to allow UCM6xxx to monitor the status of each other sending regularly SIP OPTIONS to check if it's still online.

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

Enable Heartbeat Detection

Click **Save** and **Apply Changes** to apply configuration.

Once the trunk has been created and Enable Heartbeat Detection is set, users can view the status of the peered trunk by navigating to the **Status** page.



Peer Trunk Status

Outbound Routes Configuration

On the main office UCM6xxx web GUI under "**Web UI → Extension/Trunk → Outbound Routes**", click on **+ Add**. This would allow the extensions on main office to reach extensions on Branch Office.

Configure using following parameters:

1. **Calling Rule Name:** This is for reference purposes so we choose to use "toBranch".
2. **Pattern:** The pattern used in this example is _2XXX since the Branch Office is using extension range 2XXX.
3. **Privilege Level:** Configured as "Internal". User can change it to another privilege depending on the use case.
4. **User Trunk:** Select the SIP Trunk toBranch.
5. Click on **Save** then **Apply Changes**.

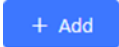
In figure 15, the pattern “_2XXX” means that whenever a user on the main office call an extension that starts with 2 and have 4 digits, this trunk will be used for the call.

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

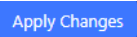
PEER Trunk – Create New Outbound Rule

- These steps will also apply when configuring the outbound route from Branch Office UCM6xxx. The pattern used will be _1XXX since the extension range on the Main Office is 1XXX.
- Users also could create Time based Condition to use Outbound Routes on a specific time, please refer to TIME CONDITION for more details.

Inbound Routes Configuration

To create an inbound rule, on the UCM webGUI, under “**Web UI → Extension/trunk → Inbound Routes**” click on  button.

For this example, Main Office UCM6xxx inbound rule needs to be configured so that when an extension on the Branch Office dials, it will be routed to the specified user. Configure using following parameters:

1. **Trunks:** Select the **toBranch** trunk.
2. **DID Pattern:** Enter in the pattern “_1XXX” | “_2XXX” as shown in below figure.
3. **DID Destination:** By DID.
4. Click on  then .

Create Inbound Rule

With this inbound rule configured, if the Main Office UCM6xxx receives a call from any extension with a leading 2 and containing 4 digits to any extension with a leading 1 and contains 4 digits it will be redirected by DID to corresponding extension.

- Those steps will also need to be applied to the Branch Office.
- Users also could create Time based Conditions to use Inbound Routes on specific times, please refer to [TIME CONDITION] for more details.

Time Condition

Users can set time condition on Inbound/Outbound rules in order to use each route on a specific time.

Supposing that Time Conditions are already set, if more details are needed, please refer to the following guide: <https://documentation.grandstream.com/knowledge-base/managing-office-time-holidays/>

Users can setup time conditions for trunks by following below steps:

1. Access the **UCM6xxx Web UI → Extension / Trunk → Outbound Routes**.
2. Navigate to time condition section.
3. Select the time condition when to use this trunk. (Office Time in this example)
4. Click on "Add" to add the time condition.

Edit Outbound Rule: Rule_2

Fallover Trunk

[+ Add](#)

TRUNKS	STRIP	PREPEND	OPTIONS
No Data			

Time Condition

* Time Condition Mode:

* Trunk:

Time Condition:

[Save](#) [Cancel](#)

TRUNKS	TIME CONDITION	TIME	WEEK	MONTH	DAY	OPTIONS
No Data						

Copyright © Grandstream Networks, Inc. 2022. All Rights Reserved.

Outbound Time Office

The example above shows that on the Office Time, calls will be made using Provider_1 trunk.

Edit Outbound Rule: Provider_1

[Cancel](#) [Save](#)

General

* Calling Rule Name:

* Pattern:

PIN Groups:

Password:

Disable This Route: ☐

Privilege Level:

PIN Groups with Privilege Level: ☐

Warning: Setting privilege level at "Internal" has potential security risks.

Enable Filter on Source Caller ID

Enable Filter on Source Caller ID: ☐

Outbound Route CID:

Call Duration Limit

Call Duration Limit: ☐

Main Trunk

* Trunk:

Strip:

Prepend:

Fallover Trunk

[+ Add](#)

TRUNKS	STRIP	PREPEND	OPTIONS
SIP Trunks -- Provider_1	2		

Time Condition

* Time Condition Mode:

[Add](#)

TRUNKS	TIME CONDITION	TIME	WEEK	MONTH	DAY	OPTIONS
SIP Trunks -- Provider_2	Out of Office Time	Default	Default	Default	Default	

Out Of Office Time

The example above shows that on Out of Office Time, calls will be made using Provider_2 trunk.

Failover Trunk

Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down.

UCM6xxx will use Failover trunks in following scenarios:

- No response from first trunk after 32 seconds.
- If UCM6xxx receives 403/407/408/503/603 SIP responses from primary trunk.
- If primary trunk is disabled.
- If primary trunk is an analog trunk and it's busy or not connected.

Users can setup failover for peer and register trunks by following below steps:

1. Go to **PBX->Basic/Call Routes->Outbound Routes**.
2. Click on **"Click to add failover trunk"**.
3. Select then the trunk to be set as failover as shown in the below figure.
4. Strip and prepend digits if needed.

Adding Failover Trunk

Click on **+ Add** to add another failover trunk.

Click on **↻** **↺** **↻** **↺** to arrange the order. If there are multiple trunks set as failovers.

Click on **✎** to edit, or **✖** to delete the Trunk from Failover list.