



Grandstream Networks, Inc.

UCM6XXX Busy Camp-on Guide



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OVERVIEW

Busy Camp-on/Call Completion is a feature that the PBX will camp on a called party and inform the caller as soon as the called party becomes available given the previous attempted call cannot be successfully established.

If a call fails to be established, the caller could request the UCM6XXX to camp on the called party by dialing the call completion request code. Then the UCM6XXX will give a call to the caller as soon as the called party becomes available. By answering the call from UCM6XXX, a call from the caller to the called party will be initiated automatically by the UCM6XXX to complete the call.

The call completion can be configured for individual extensions as well as SIP register trunks.

- When call completion is configured for individual extensions, the specific extension will get notified to complete the call when the called extension is available.
- When call completion is configured for SIP register/peer trunks, any extension in one UCM6XXX will get notified to complete the call when the called extension in the peer UCM6XXX is available if the extension has call completion configured too.

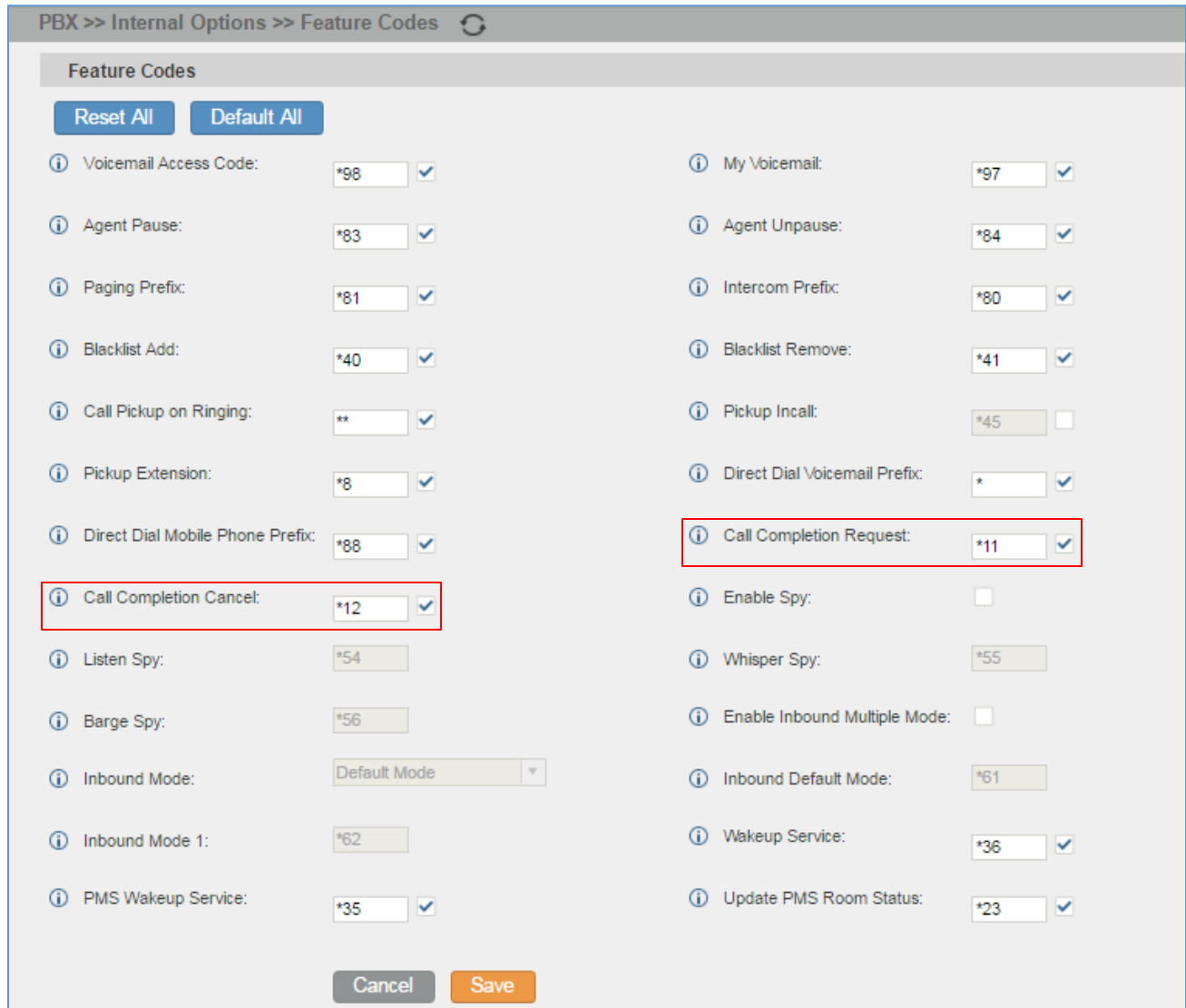
This document describes how to configure call completion for the above two applications.

Note: UCM6XXX series include UCM6100 series (UCM6102, UCM6104, UCM6108 and UCM6116), UCM6200 series (UCM6202, UCM6204 and UCM6208) and UCM6510.



CALL COMPLETION FEATURE CODE

Feature code for call completion request can be modified on web GUI->**PBX->Internal Options->Feature Codes** page. The default setting is *11 for “Call Completion Request” and *12 for “Call Completion Cancel”.



Feature Codes

Reset All Default All

Voicemail Access Code:	*98	<input checked="" type="checkbox"/>	My Voicemail:	*97	<input checked="" type="checkbox"/>
Agent Pause:	*83	<input checked="" type="checkbox"/>	Agent Unpause:	*84	<input checked="" type="checkbox"/>
Paging Prefix:	*81	<input checked="" type="checkbox"/>	Intercom Prefix:	*80	<input checked="" type="checkbox"/>
Blacklist Add:	*40	<input checked="" type="checkbox"/>	Blacklist Remove:	*41	<input checked="" type="checkbox"/>
Call Pickup on Ringing:	**	<input checked="" type="checkbox"/>	Pickup Incall:	*45	<input type="checkbox"/>
Pickup Extension:	*8	<input checked="" type="checkbox"/>	Direct Dial Voicemail Prefix:	*	<input checked="" type="checkbox"/>
Direct Dial Mobile Phone Prefix:	*88	<input checked="" type="checkbox"/>	Call Completion Request:	*11	<input checked="" type="checkbox"/>
Call Completion Cancel:	*12	<input checked="" type="checkbox"/>	Enable Spy:		<input type="checkbox"/>
Listen Spy:	*54	<input type="checkbox"/>	Whisper Spy:	*55	<input type="checkbox"/>
Barge Spy:	*56	<input type="checkbox"/>	Enable Inbound Multiple Mode:		<input type="checkbox"/>
Inbound Mode:	Default Mode	<input type="checkbox"/>	Inbound Default Mode:	*61	<input type="checkbox"/>
Inbound Mode 1:	*62	<input type="checkbox"/>	Wakeup Service:	*36	<input checked="" type="checkbox"/>
PMS Wakeup Service:	*35	<input checked="" type="checkbox"/>	Update PMS Room Status:	*23	<input checked="" type="checkbox"/>

Cancel Save

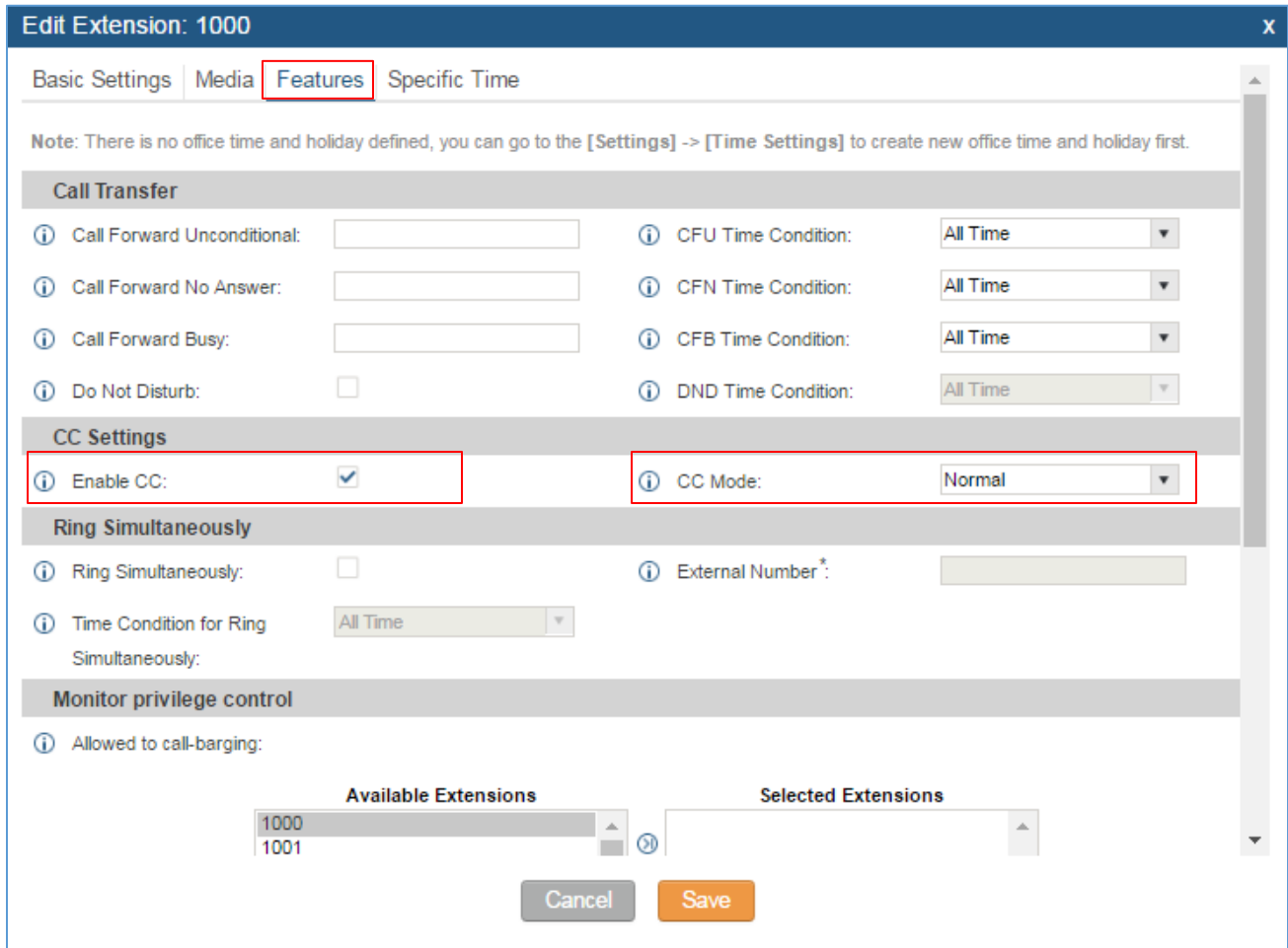
Figure 1: Call Completion Feature Code



CALL COMPLETION FOR LOCAL EXTENSIONS

Configuration

1. On UCM6XXX web GUI->**PBX->Basic/Call Routes->Extensions** page, create or edit an extension (e.g., 1000) to bring up the dialog in below figure.
2. Click on “Features” tab and make sure the following are configured:
 - “Enable CC”: selected
 - “CC Mode”: set to “Normal”



Edit Extension: 1000

Basic Settings | Media | **Features** | Specific Time

Note: There is no office time and holiday defined, you can go to the [Settings] -> [Time Settings] to create new office time and holiday first.

Call Transfer

Call Forward Unconditional: CFU Time Condition: All Time

Call Forward No Answer: CFN Time Condition: All Time

Call Forward Busy: CFB Time Condition: All Time

Do Not Disturb: DND Time Condition: All Time

CC Settings

Enable CC: CC Mode: Normal

Ring Simultaneously

Ring Simultaneously: External Number*:

Time Condition for Ring Simultaneously: All Time

Monitor privilege control

Allowed to call-barging:

Available Extensions: 1000, 1001 | Selected Extensions:

Cancel Save

Figure 2: Enable Call Completion for Extensions

3. Configure the above steps to another extension 1001 if extension 1001 is the party that will be on the call with extension 1000.

Sample Application

Assuming “user A” is using UCM6XXX extension 1000, and user B is using UCM6XXX extension 1001. Both extensions have “Enable CC” selected and “CC Mode” set to “Normal” as mentioned above.



1. Extension 1000 calls extension 1001.
2. The call fails to be established due to the following possible reasons:
 - a) Extension 1001 is busy, e.g., talking on the phone.
 - b) Extension 1001 rejects the call or the call goes to timeout.
3. At this time, extension 1000 dials “Call Completion Request” code (*11 by default) to activate camp on feature. Please note “Enable CC” option must be selected and “CC Mode” must be set to “Normal” for both extensions 1000 and 1001. Otherwise the user is not allowed to dial the call completion request code.
4. Once extension 1001 becomes available, UCM6XXX will call extension 1000. Extension 1000 has to answer the call. The following conditions for extension 1001 are considered as available:
 - a) If extension 1001 was busy when 1000 called 1001, 1001 is considered as available after the previously active call hangs up.
 - b) If extension 1001 rejected the call or the call went to timeout when 1000 called 1001, 1001 is considered as available after a new call is completed. This means extension 1001 has to initiate a new call or answer another incoming call and the new call hangs up. Otherwise, the UCM6XXX will not know whether extension 1001 is available or not.
5. A call will be initiated to extension 1001 to establish call between 1000 and 1001.

CALL COMPLETION FOR TRUNKS

Configuration

Call completion for trunks is applicable to SIP register trunks and SIP peer trunks. Two UCM6XXXs must be first configured with SIP trunks to each other. For the sake of the following illustration, we name the two UCM6XXXs involved in this example UCM1 with IP address 192.168.5.250 and UCM2 with IP address 192.168.5.143 respectively.

Using SIP Register Trunks

1. On UCM1, create extension 1000. This extension is for UCM2 to register SIP trunk to UCM1.
2. On UCM1 extension 1000, go to “Features” tab and make sure the following are configured:
 - “Enable CC”: selected
 - “CC Mode”: set to “For Trunk”

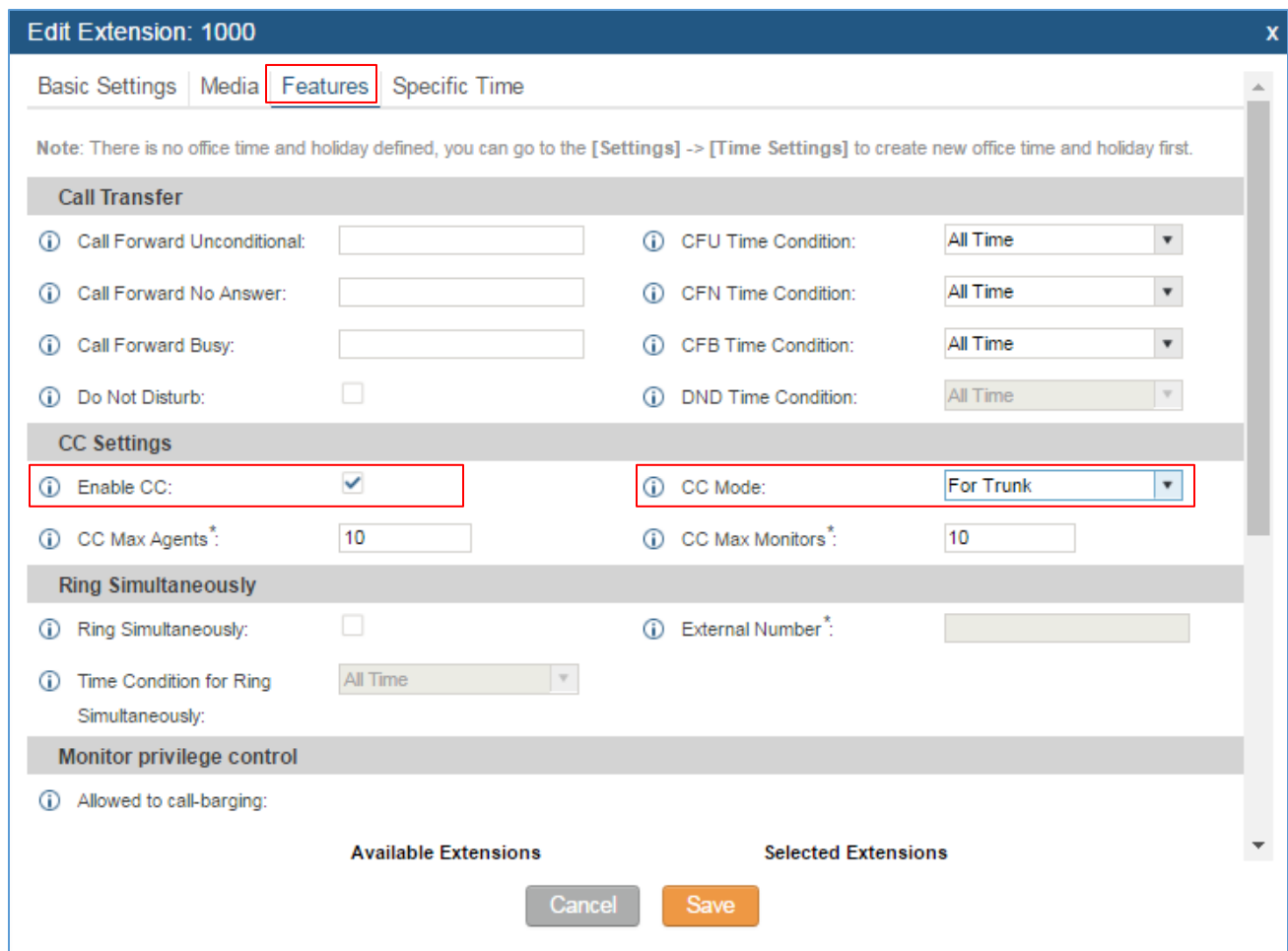
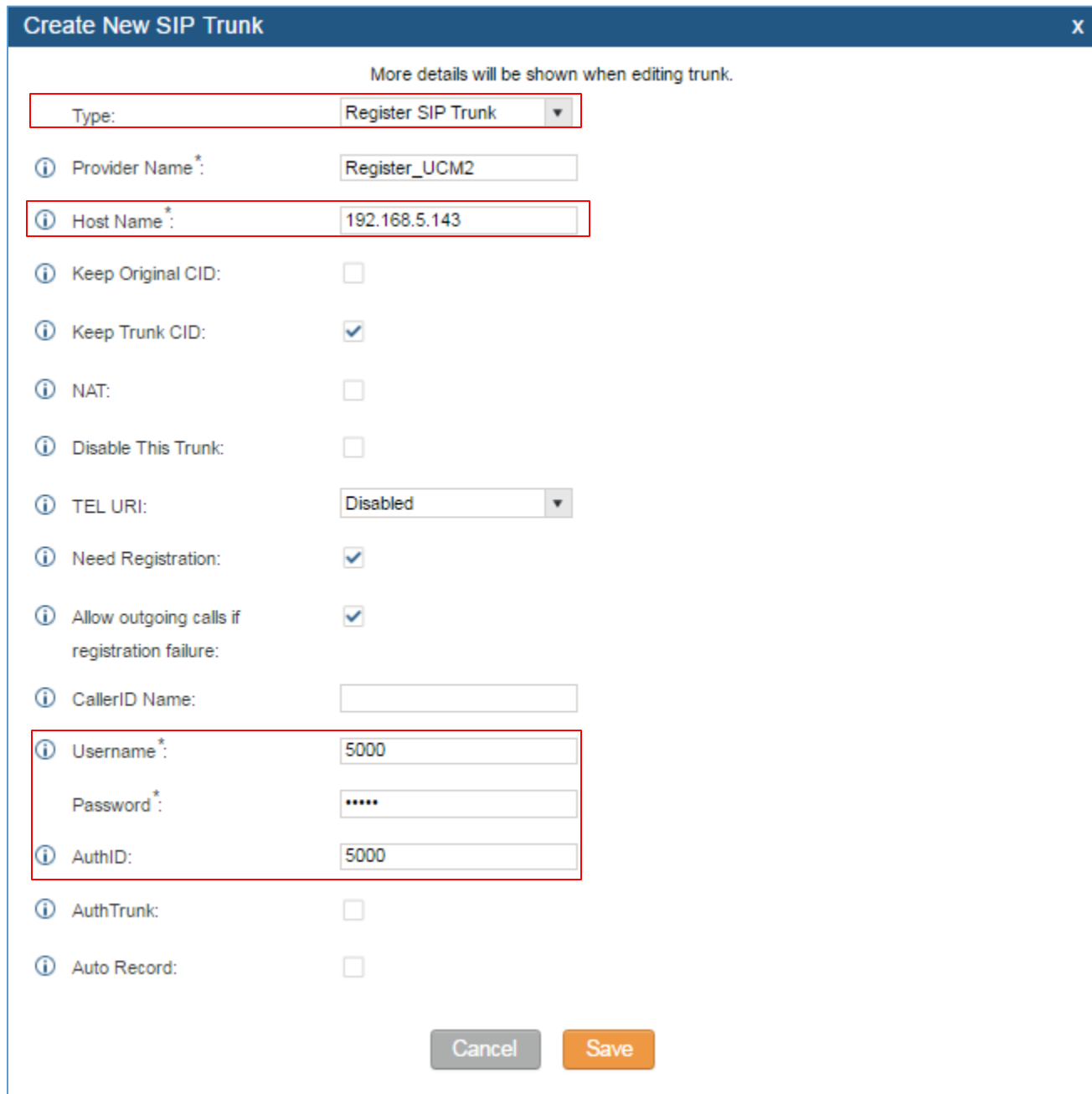


Figure 3: Enable Call Completion for SIP Register Trunk



3. Make the same configuration for extension 5000 on UCM2. This extension is for UCM1 to register SIP trunk on UCM2.
4. On UCM1, create a SIP register trunk and register to the extension 5000 on UCM2. This can be done by clicking **Create New SIP Trunk** on web GUI->**PBX->Basic/Call Routes->VoIP Trunks**. The following figure shows the configuration for new SIP trunk on UCM1.



More details will be shown when editing trunk.

Type: Register SIP Trunk

Provider Name*: Register_UCM2

Host Name*: 192.168.5.143

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Need Registration:

Allow outgoing calls if registration failure:

CallerID Name:

Username*: 5000

Password*: *****

AuthID: 5000

AuthTrunk:

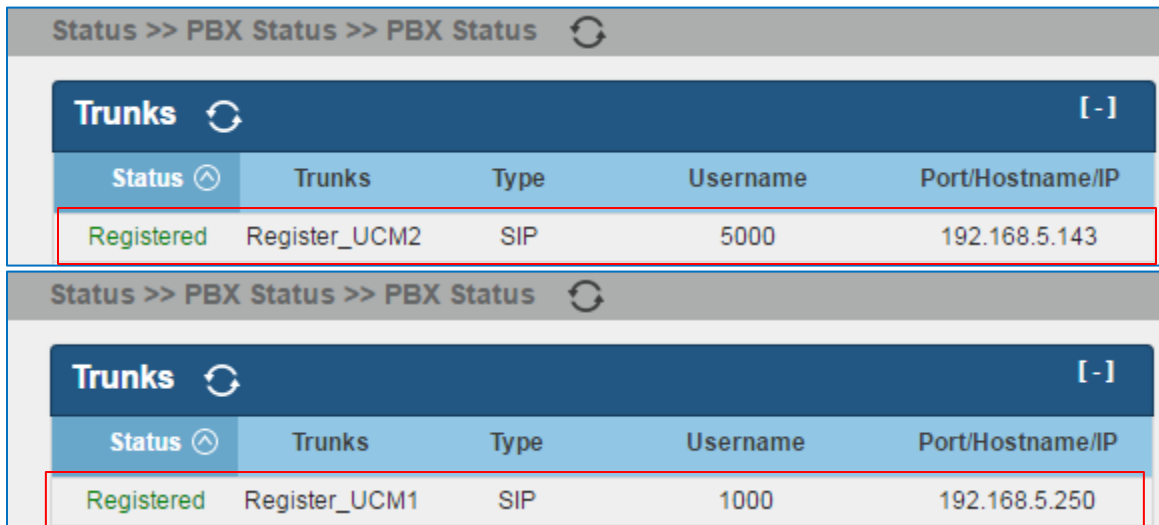
Auto Record:

Cancel Save

Figure 4: Create SIP Register Trunk

- **Type:** Select “Register SIP Trunk”.
- **Host Name:** Enter the IP address of the UCM to register to.

- **Username:** The extension number on the UCM to register to.
 - **AuthID:** Same as Username.
 - **Password:** The password of the extension number on the UCM to register to.
5. Similar to step 4, on UCM2, create a SIP register trunk and register to the extension 1000 on UCM1.
 6. Check the registration status of the trunks on web GUI->**Status->PBX Status**. If configured successfully, the status for the trunk should show as “Registered”.



Status	Trunks	Type	Username	Port/Hostname/IP
Registered	Register_UCM2	SIP	5000	192.168.5.143

Status	Trunks	Type	Username	Port/Hostname/IP
Registered	Register_UCM1	SIP	1000	192.168.5.250

Figure 5: SIP Register Trunk Status

7. Configure inbound and outbound rules on two UCMs to make sure the extensions on UCM1 can reach the extensions on UCM2 through the SIP register trunk and vice versa.
8. For the extensions on both UCM6XXX that you would like to use call completion, go to the UCM6XXX web GUI->**PBX->Basic/Call Routes->Extensions** page, create or edit extension with the following configured in “Features” tab:
 - “Enable CC”: selected
 - “CC Mode”: set to “Normal”

Now, Call Completion feature for trunks is ready to be used when making calls between the 2 UCM6XXX extensions.

Using SIP Peer Trunks

1. On UCM1, create a SIP peer trunk with UCM2. This can be done by clicking Create New SIP Trunk on web GUI->**PBX->Basic/Call Routes->VoIP Trunks**. The following figure shows the configuration for new SIP trunk on UCM1.

Create New SIP Trunk
X

More details will be shown when editing trunk.

Type: Peer SIP Trunk

Provider Name *

Host Name *

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Caller ID:

CallerID Name:

Auto Record:

Cancel
Save

Figure 6: Create SIP Peer Trunk

- Type:** Select “Register SIP Trunk”.
- Host Name:** Enter the IP address of the UCM to register to.

1.1. After saving, press edit button as shown in figure below:

PBX >> Basic/Call Routes >> VoIP Trunks ↻

VoIP Trunks

Create New SIP Trunk
Create New IAX Trunk
View: 10

Provider Name	Technology	Type	Hostname/IP	Username	Options
Peer_UCM2	SIP	peer	192.168.5.143		✎ ☎ 🏠 🗑️

Figure 7: Edit SIP Peer Trunk

1.2. Access “Advanced Settings” tab and set following options:

- “Enable Heartbeat Detection”: selected. This setting is optional, if activated it will help to check the status of the trunk.
- “Enable CC”: selected.



Edit SIP Trunk: Peer_UCM2
X

Basic Settings
Advanced Settings

① Codec Preference:

① Send PPI Header:

① Send PAI Header:

① DID Mode: Request-line

① DTMF Mode: Default

① Enable Heartbeat Detection:

① Heartbeat Frequency*: 60

① The Maximum Number of Call Lines*: 0

① Fax Mode: None

① SRTP: Disabled

① Sync LDAP Enable:

Available Codecs		Selected Codecs
G.722	⊕	PCMU
AAL2-G.726-32	⊖	PCMA
ADPCM	⊕	GSM
G.723	⊖	G.726
H.263	⊕	G.729

CC Settings

① Enable CC:

① CC Max Agents*: 10

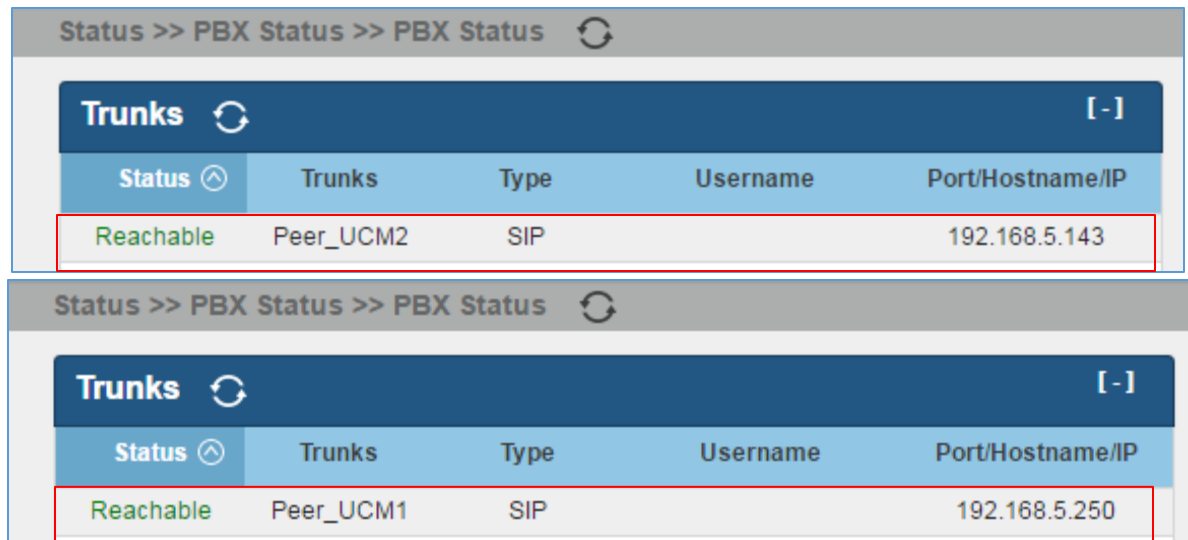
① CC Max Monitors*: 10

Cancel
Save

Figure 8: SIP Peer Trunk – Advanced Settings

2. Similar to step 1, on UCM2, create a SIP peer trunk with UCM1.
3. Check the trunks status on web GUI->**Status**->**PBX Status**. If configured successfully, the status for the trunk should show as "Reachable".





Status >> PBX Status >> PBX Status

Status	Trunks	Type	Username	Port/Hostname/IP
Reachable	Peer_UCM2	SIP		192.168.5.143

Status >> PBX Status >> PBX Status

Status	Trunks	Type	Username	Port/Hostname/IP
Reachable	Peer_UCM1	SIP		192.168.5.250

Figure 9: SIP Peer Trunk Status

- Configure inbound and outbound rules on two UCMs to make sure the extensions on UCM1 can reach the extensions on UCM2 through the SIP peer trunk and vice versa.
- For the extensions on both UCM6XXX that you would like to use call completion, go to the UCM6XXX web GUI->PBX->Basic/Call Routes->Extensions page, create or edit extension with the following configured in “Features” tab:
 - “Enable CC”: selected
 - “CC Mode”: set to “Normal”

Now, Call Completion feature for trunks is ready to be used when making calls between the 2 UCM6XXX extensions.

Sample Application

After the above configuration, assuming user A is using extension 1005 on UCM1 and user B is using extension 5001 on UCM2.

- Extension 1005 on UCM1 calls extension 5001 on UCM2.
- The call fails to be established due to the following possible reasons:
 - Extension 5001 is busy, e.g., talking on the phone.
 - Extension 5001 rejects the call or the call goes to timeout.
- At this time, extension 1005 dials “Call Completion Request” code (*11 by default) to activate camp on feature. Please make sure “Enable CC” option is enabled and “CC Mode” is set to “Normal” for both extension 1005 and extension 5001. Otherwise, the user is not allowed to dial the call completion request code.
- Once extension 5001 becomes available, UCM6XXX will call extension 1005. Extension 1005 has to answer the call. The following conditions for extension 5001 are considered as available.



- a) If extension 5001 was busy when 1005 called 5001, 5001 is considered as available after the previously active call hangs up.
 - b) If extension 5001 rejected the call or the call went to timeout when 1005 called 5001, 5001 is considered as available after a new call is completed. This means extension 5001 has to initiate a new call or answer another incoming call and the new call hangs up. Otherwise the UCM6XXX will not know whether extension 5001 is available or not.
7. A call will be initiated to extension 1005 to establish call between 1005 and 5001.

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