



# Grandstream Networks, Inc.

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## UCM6XXX Busy Camp-on Guide



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## OVERVIEW

Busy Camp-on/Call Completion is a feature where 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.

When trying to reach an extension which is already busy, 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/peer 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

The feature code for call completion request can be modified on **web GUI** → **Call Features** → **Feature Codes** page. The default setting is \*11 for “Call Completion Request” and \*12 for “Call Completion Cancel”.

The screenshot shows the 'Feature Codes' configuration page with the following settings:

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

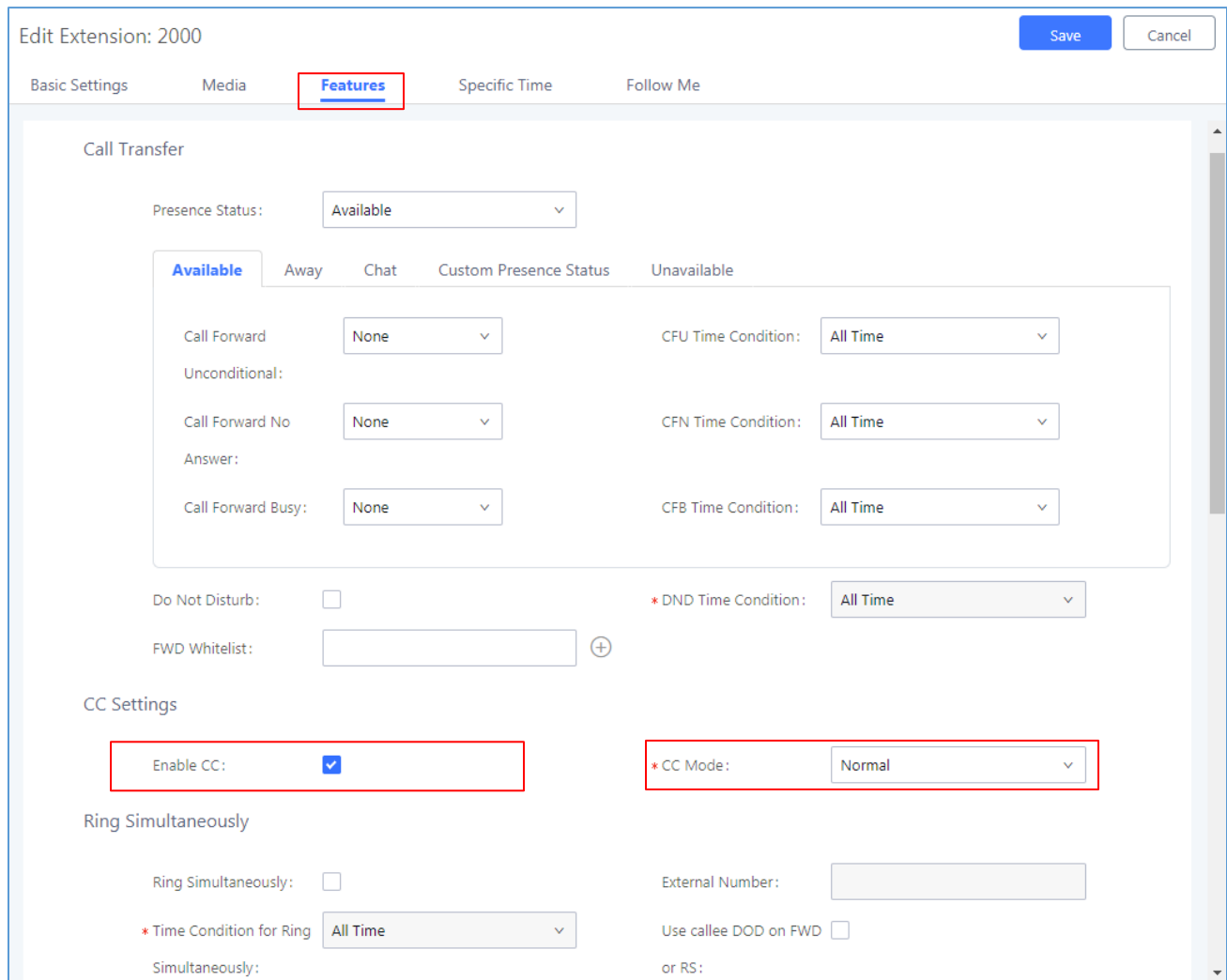
Figure 1: Call Completion Feature Code



# CALL COMPLETION FOR LOCAL EXTENSIONS

## Configuration

1. On UCM6XXX web GUI → **Extensions/Trunk** → **Extensions** page, create or edit an extension (e.g., 2000) 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: 2000 Save Cancel

Basic Settings   Media   **Features**   Specific Time   Follow Me

**Call Transfer**

Presence Status: Available

Available   Away   Chat   Custom Presence Status   Unavailable

Call Forward: None   CFU Time Condition: All Time

Unconditional:

Call Forward No Answer: None   CFN Time Condition: All Time

Call Forward Busy: None   CFB Time Condition: All Time

Do Not Disturb:    \* DND Time Condition: All Time

FWD Whitelist:

**CC Settings**

Enable CC:    \* CC Mode: Normal

**Ring Simultaneously**

Ring Simultaneously:    External Number:

\* Time Condition for Ring: All Time   Use callee DOD on FWD:

Simultaneously:   or RS:

**Figure 2: Enable Call Completion for Extensions**

3. Configure the above steps to another extension 2001 if extension 2001 is the party that will be on the call with extension 2000.

## Sample Application

Assuming “user A” is using UCM6XXX extension 2000, and user B is using UCM6XXX extension 2001. Both extensions have “Enable CC” selected and “CC Mode” set to “Normal” as mentioned above.

1. Extension 2000 calls extension 2001.
2. The call fails to be established due to the following possible reasons:
  - a) Extension 2001 is busy, e.g., talking on the phone.
  - b) Extension 2001 rejects the call or the call goes to timeout.
3. At this time, extension 2000 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 2000 and 2001. Otherwise the user is not allowed to dial the call completion request code.
4. Once extension 2001 becomes available, UCM6XXX will call extension 2000. Extension 2000 has to answer the call. The following conditions for extension 2001 are considered as available:
  - a) If extension 2001 was busy when 2000 called 2001, 1001 is considered as available after the previously active call hangs up.
  - b) If extension 2001 rejected the call or the call went to timeout when 2000 called 2001, 2001 is considered as available after a new call is completed. This means extension 2001 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 2001 is available or not.
5. A call will be initiated to extension 2001 to establish call between 2000 and 2001.



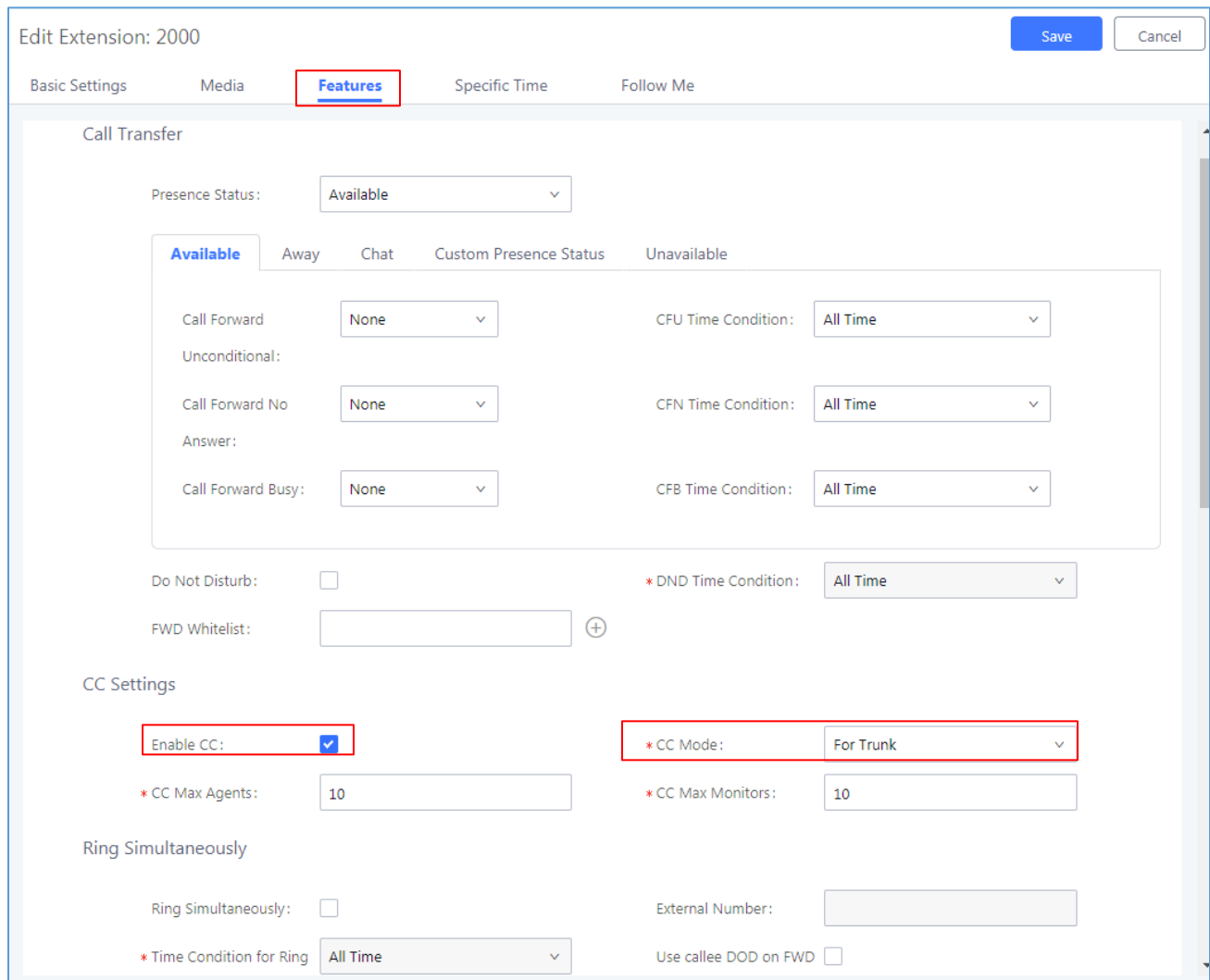
# CALL COMPLETION FOR TRUNKS

## 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 192.168.6.133 and UCM2 with IP address 192.168.5.143 respectively.

### Using SIP Register Trunks

1. On UCM1, create extension 2000. This extension is for UCM2 to register SIP trunk to UCM1.
2. On UCM1 extension 2000, go to “Features” tab and make sure the following are configured:
  - “Enable CC”: selected
  - “CC Mode”: set to “For Trunk”

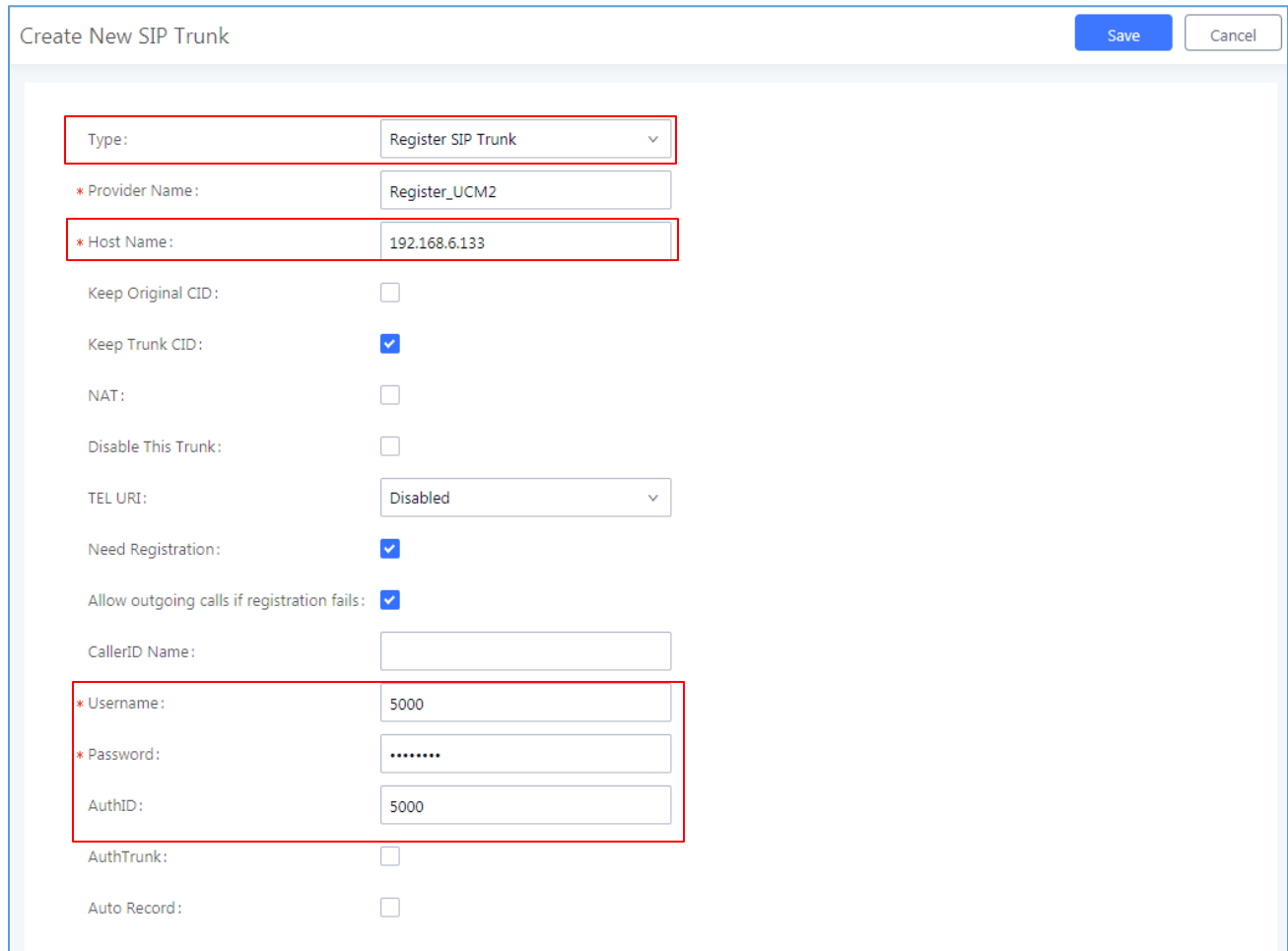


The screenshot shows the configuration interface for extension 2000. The 'Features' tab is selected and highlighted with a red box. Under the 'Call Transfer' section, various call forwarding options are set to 'None' and time conditions are set to 'All Time'. In the 'CC Settings' section, 'Enable CC' is checked (highlighted with a red box) and '\* CC Mode' is set to 'For Trunk' (also highlighted with a red box). Other settings like 'CC Max Agents' and 'CC Max Monitors' are set to 10.

**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** → **Extension/Trunk** → **VoIP Trunks**. The following figure shows the configuration for new SIP trunk on UCM1.



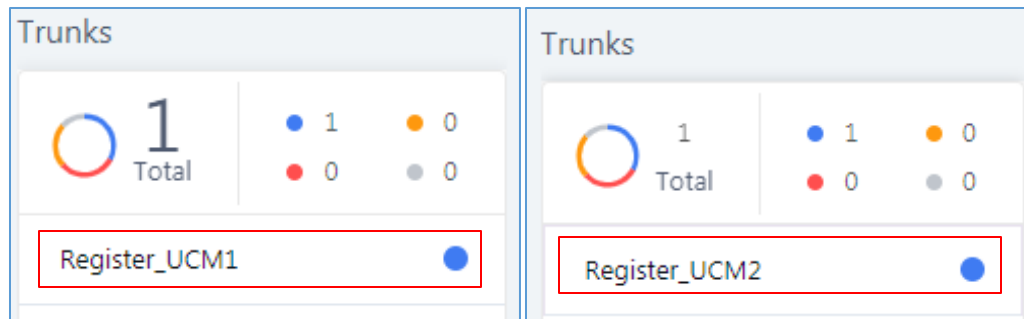
The screenshot shows the 'Create New SIP Trunk' configuration window. The form contains the following fields and values:

- Type: Register SIP Trunk (dropdown menu)
- \* Provider Name: Register\_UCM2
- \* Host Name: 192.168.6.133
- Keep Original CID:
- Keep Trunk CID:
- NAT:
- Disable This Trunk:
- TEL URI: Disabled (dropdown menu)
- Need Registration:
- Allow outgoing calls if registration fails:
- CallerID Name: (empty text field)
- \* Username: 5000
- \* Password: (masked with dots)
- AuthID: 5000
- AuthTrunk:
- Auto Record:

**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 6000 on UCM1.
  6. Check the registration status of the trunks on **web GUI**→**System Status** → **Dashboard**. If configured successfully, the status for the trunk should show as “Registered”.





**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** → **Extension / Trunk** → **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** → **Extension/Trunk** → **VoIP Trunks**. The following figure shows the configuration for new SIP trunk on UCM1.

Create New SIP Trunk Save Cancel

Type:	Peer SIP Trunk
* Provider Name:	Peer_UCM2
* Host Name:	192.168.6.133
Keep Original CID:	<input type="checkbox"/>
Keep Trunk CID:	<input type="checkbox"/>
NAT:	<input type="checkbox"/>
Disable This Trunk:	<input type="checkbox"/>
TEL URI:	Disabled
Caller ID:	<input type="text"/>
CallerID Name:	<input type="text"/>
Auto Record:	<input type="checkbox"/>





**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:

VoIP Trunks

+ Create New SIP Trunk + Create New IAX Trunk

Provider Name	Terminal Type	Type	Hostname/IP	Username	Options
Peer_UCM2	SIP	peer	192.168.6.133		   

**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 Save Cancel

Basic Settings Advanced Settings

Codec Preference:

Available Codec		Selected Codec
<input type="checkbox"/> G.722	< >	<input type="checkbox"/> PCMU
<input type="checkbox"/> AAL2-G...	↑ ↓	<input type="checkbox"/> PCMA
<input type="checkbox"/> ADPCM		<input type="checkbox"/> GSM
<input type="checkbox"/> G.723		<input type="checkbox"/> G.726
<input type="checkbox"/> H.263		<input type="checkbox"/> G.729

Send PPI Header:

Send PAI Header:

DID Mode: Request-line

DTMF Mode: Default

**Enable Heartbeat Detection:**

\* Heartbeat Frequency: 60

\* The Maximum Number of Call Lines: 0

Fax Mode: None

SRTP: Disabled

Sync LDAP Enable:

**CC Settings**

**Enable CC:**

\* CC Max Agents: 10

\* CC Max Monitors: 10

Figure 8: SIP Peer Trunk – Advanced Settings

- Similar to step 1, on UCM2, create a SIP peer trunk with UCM1.
- Check the trunks status on **web GUI** → **System Status** → **Dashboard**. If configured successfully, the status for the trunk should show as “Reachable”.

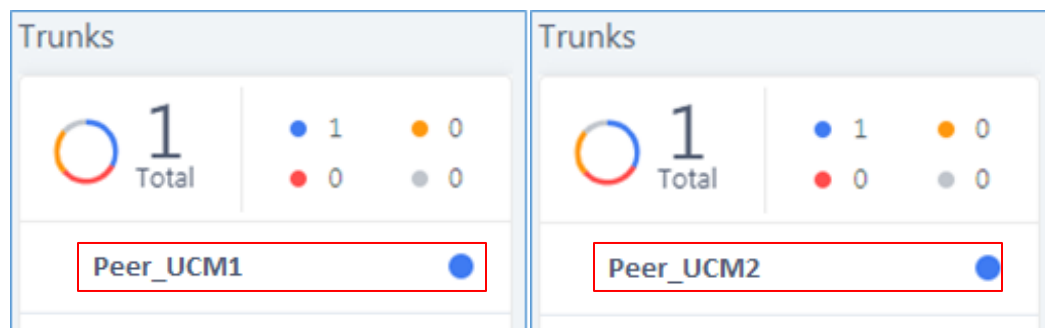


Figure 9: SIP Peer Trunk Status

- 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.



5. For the extensions on both UCM6XXX that you would like to use call completion, go to the UCM6XXX **web GUI → Extension/Trunk → 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.

1. Extension 1005 on UCM1 calls extension 5001 on UCM2.
2. The call fails to be established due to the following possible reasons:
  - a) Extension 5001 is busy, e.g., talking on the phone.
  - b) Extension 5001 rejects the call or the call goes to timeout.
3. 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.
6. 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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