



FIBERME Communications LLC.

FCM630A Series IP PBX - 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 FCM630A to camp on the called party by dialing the call completion request code. Then the FCM630A will give a call to the caller as soon as the called party becomes available. By answering the call from FCM630A, a call from the caller to the called party will be initiated automatically by the FCM630A 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 FCM630A will get notified to complete the call when the called extension in the peer FCM630A is available if the extension has call completion configured too.

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



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. It includes a navigation bar with 'Feature Maps', 'DND/Call Forward', and 'Feature Codes'. Below the navigation are 'Reset All' and 'Default All' buttons. The main area contains a grid of feature codes, each with a text input field for the code and a checkbox for its status. Two entries are highlighted with red boxes: '* Call Completion Cancel' with value '*12' and '* Call Completion Request' with value '*11'. Other visible codes include Voicemail Access Code (*98), Agent Pause (*83), Paging Prefix (*81), Blacklist Add (*40), Pickup on Ringing Prefix (**), Pickup Extension (*8), Direct Dial Mobile Phone Prefix (*88), My Voicemail (*97), Agent Unpause (*84), Intercom Prefix (*80), Blacklist Remove (*41), Pickup In-call Prefix (*45), Direct Dial Voicemail Prefix (*), Enable Spy, Whisper Spy (*55), Wakeup Service (*36), Update PMS Room Status (*23), Dynamic Agent Logout (*85), Listen Spy (*54), Barge Spy (*56), PMS Wakeup Service (*35), Presence Status (*48), and Voicemail Group Access Code (*99).

Feature Code	Value	Status
* Voicemail Access Code	*98	<input checked="" type="checkbox"/>
* Agent Pause	*83	<input checked="" type="checkbox"/>
* Paging Prefix	*81	<input checked="" type="checkbox"/>
* Blacklist Add	*40	<input checked="" type="checkbox"/>
* Pickup on Ringing Prefix	**	<input checked="" type="checkbox"/>
* Pickup Extension	*8	<input checked="" type="checkbox"/>
* Direct Dial Mobile Phone Prefix	*88	<input checked="" type="checkbox"/>
* Call Completion Cancel	*12	<input checked="" type="checkbox"/>
* Listen Spy	*54	<input type="checkbox"/>
* Barge Spy	*56	<input type="checkbox"/>
* PMS Wakeup Service	*35	<input checked="" type="checkbox"/>
* Presence Status	*48	<input checked="" type="checkbox"/>
* Voicemail Group Access Code	*99	<input checked="" type="checkbox"/>
* 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 Prefix	*45	<input type="checkbox"/>
* Direct Dial Voicemail Prefix	*	<input checked="" type="checkbox"/>
* Call Completion Request	*11	<input checked="" type="checkbox"/>
Enable Spy		<input type="checkbox"/>
* Whisper Spy	*55	<input type="checkbox"/>
* Wakeup Service	*36	<input checked="" type="checkbox"/>
* Update PMS Room Status	*23	<input checked="" type="checkbox"/>
* Dynamic Agent Logout	*85	<input checked="" type="checkbox"/>

Figure 1: Call Completion Feature Code



CALL COMPLETION FOR LOCAL EXTENSIONS

Configuration

1. On FCM630A **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”

The screenshot shows the 'Edit Extension: 1000' interface. At the top, there are tabs for 'Basic Settings', 'Media', 'Features' (highlighted with a red box), 'Specific Time', and 'Follow Me'. Below the tabs, there are 'Cancel' and 'Save' buttons. The main content area is divided into sections. The 'Call Transfer' section includes a 'Presence Status' dropdown set to 'Available'. Below this are sub-tabs for 'Available', 'Away', 'Chat', 'Custom Presence Status', and 'Unavailable'. The 'Available' sub-tab is active and contains several settings: 'Call Forward Unconditional', 'Call Forward No Answer', and 'Call Forward Busy' are all set to 'None'; 'CFU Time Condition', 'CFN Time Condition', and 'CFB Time Condition' are all set to 'All Time'. There is also a 'Do Not Disturb' checkbox (unchecked) and a '* DND Time Condition' dropdown set to 'All Time'. A 'FWD Whitelist' field is empty, with an 'Add FWD Whitelist' button below it. The 'CC Settings' section at the bottom has 'Enable CC' checked (highlighted with a red box) and '* CC Mode' set to 'Normal' (highlighted with a red box).

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 FCM630A extension 2000, and user B is using FCM630A 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, FCM630A 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 FCM630A 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 FCM630As must be first configured with SIP trunks to each other. For the sake of the following illustration, we name the two FCM630As involved in this example FCM1 with IP address 192.168.6.133 and FCM2 with IP address 192.168.5.143 respectively.

Using SIP Register Trunks

1. On FCM1, create extension 2000. This extension is for FCM2 to register SIP trunk to FCM1.
2. On FCM1 extension 2000, go to “Features” tab and make sure the following are configured:
 - “Enable CC”: selected
 - “CC Mode”: set to “For Trunk”

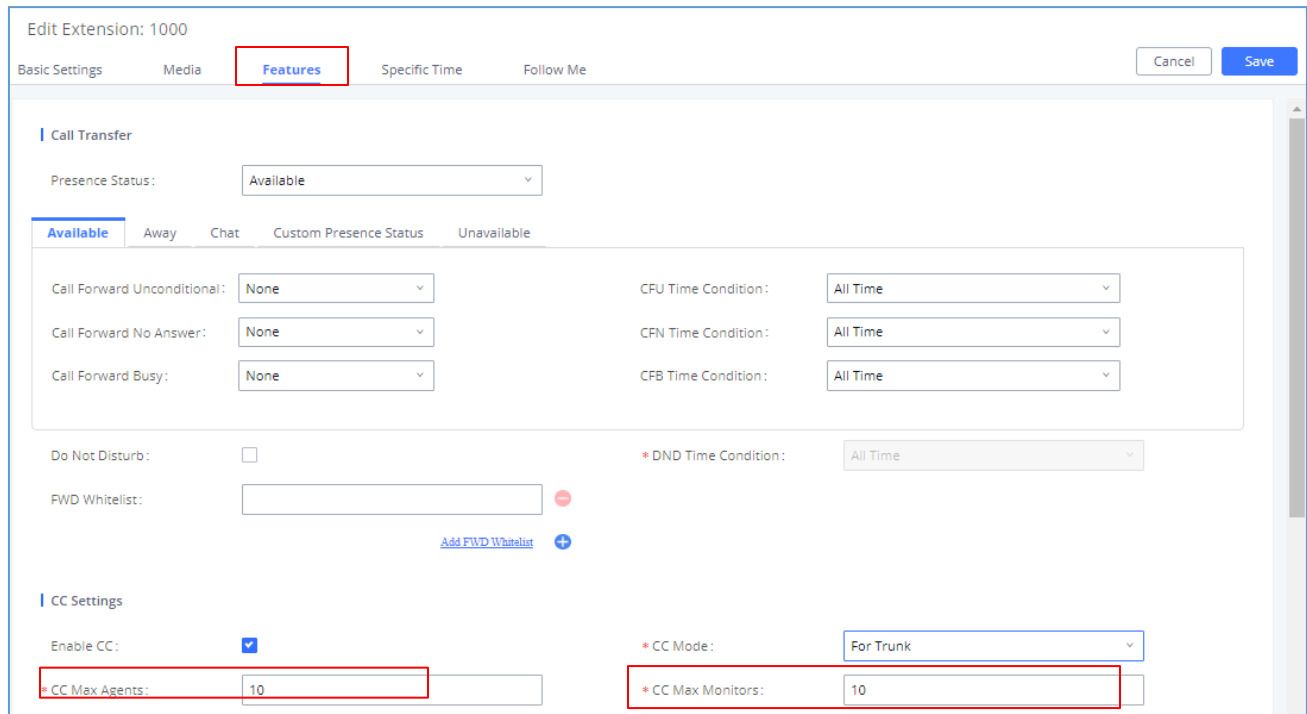


Figure 3: Enable Call Completion for SIP Register Trunk

3. Make the same configuration for extension 5000 on FCM2. This extension is for FCM1 to register SIP trunk on FCM2.
4. On FCM1, create a SIP register trunk and register to the extension 5000 on FCM2. 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 FCM1.



Create New SIP Trunk

Type: Register SIP Trunk

* Provider Name: Register_FCM2

* Host Name: 192.168.6.133

Transport: UDP

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Need Registration:

Allow outgoing calls if registration fails:

CallerID Name:

* Username: 5000

* Password:

AuthID: 5000

Figure 4: Create SIP Register Trunk

- **Type:** Select "Register SIP Trunk".
 - **Host Name:** Enter the IP address of the FCM to register to.
 - **Username:** The extension number on the FCM to register to.
 - **AuthID:** Same as Username.
 - **Password:** The password of the extension number on the FCM to register to.
5. Similar to step 4, on FCM2, create a SIP register trunk and register to the extension 6000 on FCM1.
 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".

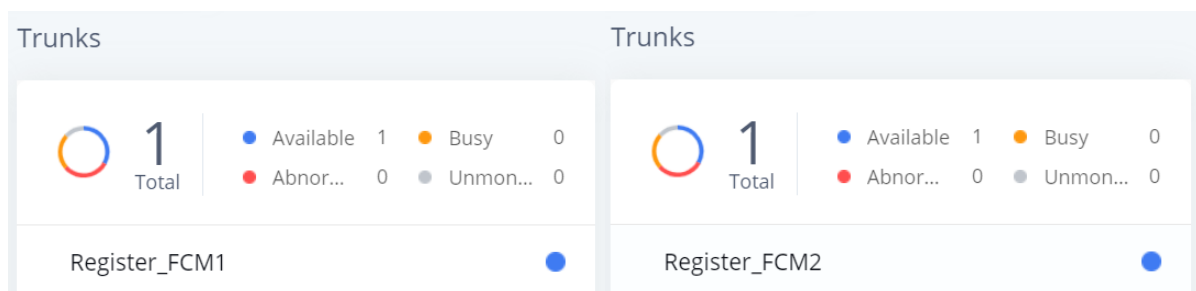


Figure 5: SIP Register Trunk Status



7. Configure inbound and outbound rules on two FCMs to make sure the extensions on FCM1 can reach the extensions on FCM2 through the SIP register trunk and vice versa.
8. For the extensions on both FCM630A that you would like to use call completion, go to the FCM630A **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 FCM6X30A extensions.

Using SIP Peer Trunks

1. On FCM1, create a SIP peer trunk with FCM2. 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 FCM1.

Create New SIP Trunk

Type:	Peer SIP Trunk ▼
* Provider Name:	Peer_FCM2
* Host Name:	192.168.6.133
Transport:	UDP ▼
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 ▼
CallerID Number:	<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 FCM to register to.

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



VoIP Trunks

VoIP Trunks Trunk Group

+ Add SIP Trunk + Add IAX Trunk





PROVIDER NAME	TERMINAL TYPE	TYPE	HOSTNAME/IP	USERNAME	OPTIONS
Peer_FCM2	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_FCM2

Basic Settings **Advanced Settings**

G.722.1

G.722.1C

G.723

H.263

PCMA

GSM

G.726

G.729

Send PPI Header:

Send PAI Header:

Passthrough PAI Header:

Send PANI Header:

Send Anonymous:

DID Mode: Request-line

DTMF Mode: Default

Enable Heartbeat Detection:

* Heartbeat Frequency (s): 60

Figure 8: SIP Peer Trunk – Advanced Settings



2. Similar to step 1, on FCM2, create a SIP peer trunk with FCM1.
3. Check the trunks status on **web GUI → System Status → Dashboard**. If configured successfully, the status for the trunk should show as “Reachable”.

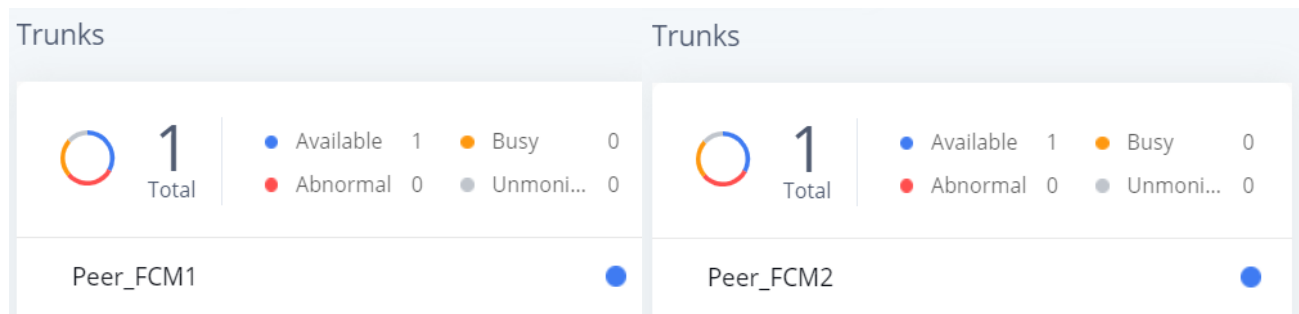


Figure 9: SIP Peer Trunk Status

4. Configure inbound and outbound rules on two FCMs to make sure the extensions on FCM1 can reach the extensions on FCM2 through the SIP peer trunk and vice versa.
5. For the extensions on both FCM630A that you would like to use call completion, go to the FCM630A **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 FCM630A extensions.

Sample Application

After the above configuration, assuming user A is using extension 1005 on FCM1 and user B is using extension 5001 on FCM2.

1. Extension 1005 on FCM1 calls extension 5001 on FCM2.
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, FCM630A 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 FCM630A 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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