



FIBERME Communications LLC.

Configuring FCM630A Series with HT813

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OVERVIEW

This document describes basic configuration to interconnect FCM630A series and HT813. This is typically applied to the scenario where users would like to add a HT813 not only as a remote extension but also as an external PSTN trunk. It could be common that we prefer to grab a PSTN line in a remote location and use the carrier service on another remote office, in this case this guide will help you implement this configuration.

There are two ways to set up the FCM630A series IP PBX with the HT813.

- **Method 1:** Register the HT813 to the FCM630A directly.
- **Method 2:** Configure HT813 as a SIP peer trunk.

The following illustration show the typical setup that will be used in this guide:



Figure 1: Typical Architecture

 **Warning:**

- When using the IVR in FCM630A, please be aware that if "Dial Trunk" option is turned on in IVR settings, the callers into the IVR will be able to dial outbound call using FCM630A's trunk. The IVR's permission level will be used when making outbound calls in this case. Please select proper permission level for the IVR to control the outbound calls allowed via "Dial Trunk".
-



METHOD 1: REGISTER HT813 TO FCM630A

Create Extension on FCM630A

On the FCM630A web GUI, create two extensions under Extension/Trunk→Extensions. These two extensions are used for HT813 FXS and FXO registration.

The password for the extension will be randomly generated if not specified.

The screenshot shows the 'Create New Extension' web interface. The 'Basic Settings' tab is active. The 'Select Extension Type' is set to 'SIP Extension' and 'Select Add Method' is set to 'Single'. Under the 'General' section, the 'Extension' field is highlighted with a red box and contains the value '1000'. The 'Permission' is set to 'Internal'. The 'Voicemail Password' is '1413097'. The 'SIP/IAX Password' field is also highlighted with a red box and contains the value 'd~HBGjj7'. Other fields include 'AuthID', 'Send Voicemail to Email' (set to 'Default'), 'Enable Keep-alive' (unchecked), 'CallerID Number', 'Voicemail' (set to 'Local Voicemail'), 'Skip Voicemail Password' (unchecked), 'Verification', 'Keep Voicemail after' (set to 'Default'), 'Emailing', and 'Keep-alive Frequency' (set to '60').

Figure 2: Create Extension 1000 on the FCM630A

The screenshot shows the 'Create New Extension' web interface for extension 1001. The 'Basic Settings' tab is active. The 'Select Extension Type' is set to 'SIP Extension' and 'Select Add Method' is set to 'Single'. Under the 'General' section, the 'Extension' field is highlighted with a red box and contains the value '1001'. The 'Permission' is set to 'Internal'. The 'Voicemail Password' is '5615081'. The 'SIP/IAX Password' field is also highlighted with a red box and contains the value 'ru^h8ByT'. Other fields include 'AuthID', 'Send Voicemail to Email' (set to 'Default'), 'Enable Keep-alive' (unchecked), 'CallerID Number', 'Voicemail' (set to 'Local Voicemail'), 'Skip Voicemail Password' (unchecked), 'Verification', 'Keep Voicemail after' (set to 'Default'), 'Emailing', and 'Keep-alive Frequency' (set to '60').

Figure 3: Create Extension 1001 on the FCM630A



Create IVR on FCM630A

On the FCM630A web GUI, create an IVR extension under **Call Features**→**IVR**. This is to receive the calls forwarded from the HT813.

In IVR settings, if "Dial Other Extensions" is enabled, the calls forwarded to the FCM630A IVR will be able to reach the internal extensions registered to the FCM630A. Also, you can assign the "Key Pressing Event" to different destinations.

The screenshot shows the 'Create New IVR' configuration page. The 'Basic Settings' tab is active. The 'Name' field is highlighted with a red box and contains the text 'HT813_IVR'. Below it, the 'Extension' field contains '7001'. The 'Dial Trunk' checkbox is unchecked. Under 'Dial Other Extensions', the 'Extension' checkbox is checked, while others are unchecked. The 'IVR Black/Whitelist' dropdown is set to 'Disable'. The 'Replace Display Name' and 'Return to IVR Menu' checkboxes are unchecked. The 'Alert-info' dropdown is set to 'None'. The 'Prompt' dropdown is set to 'welcome', with an 'Upload Audio File' button to its right. Below the prompt field is an 'Add Prompt +' button. The 'Digit Timeout' field contains '3', 'Response Timeout' contains '10', 'Response Timeout Prompt' dropdown is 'ivr-create-timeout', and 'Invalid Input Prompt' dropdown is 'invalid'. Both have 'Upload Audio File' buttons. The 'Response Timeout Prompt Repeats' dropdown is '3' and 'Invalid Input Prompt Repeats' dropdown is '3'. The 'Language' dropdown is 'Default'. 'Cancel' and 'Save' buttons are in the top right corner.

Figure 4: Create IVR 7000 on the FCM630A

Configure FXS Port on HT813

1. Connect an analog phone to the HT813 FXS port.
2. On the HT813 web GUI, go to FXS Port setting page, configure to register the FXS port to the FCM630A extension 1000. Please refer to the highlighted settings in the following figure.

In this example, the FCM630A IP address is 192.168.5.190.



Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: No Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

Allow DHCP Option 120 (override SIP server): No Yes

SIP Transport: UDP TCP TLS (default is UDP)

SIP URI Scheme When Using TLS: sip sips

Use Actual Ephemeral Port in Contact with TCP/TLS: No Yes

NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV

DNS SRV use Registered IP: No Yes

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 5: Configure FXS Port on the HT813

Configure FXO Port on HT813

1. Connect the PSTN line to the HT813 FXO port.
2. On the HT813 web GUI, go to FXO Port setting page, configure to register the FXO port to the FCM630A extension 1001. Please refer to the highlighted settings and other necessary settings in the following figures.

In this example, the FCM630A IP address is 192.168.5.190.



Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: No Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

SIP Transport: UDP TCP TLS (default is UDP)

NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV

DNS SRV use Registered IP: No Yes

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 6: Configure FXO Port on the HT813 - Registration

Since we are going to use IVR when the call is forwarded to the FCM630A, the FCM630A will need to be able to detect the DTMF digits. Configure the HT813 FXO port DTMF settings as below as an initial setup.

<i>Preferred DTMF method (in listed order):</i>	Priority 1: <input type="text" value="RFC2833"/>
	Priority 2: <input type="text" value="SIP INFO"/>
	Priority 3: <input type="text" value="In-audio"/>

Figure 7: Configure FXO Port on the HT813 - DTMF Settings

There are a few necessary changes to be made in FXO termination section and Channel Dialing section as well.



FXO Termination

Enable Current Disconnect: No Yes (Default Yes. If set to yes, enter threshold below)

Current Disconnect Threshold (ms): (50-800 milliseconds. Default 100 milliseconds)

Enable PSTN Disconnect Tone Detection: No Yes (Default No)
(If set to yes, the following tone is used as the disconnect signal)

PSTN Disconnect Tone:
(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)
(Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)
(Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)

Enable Polarity Reversal: No Yes (Default No. Check with your PSTN carrier before setting to Yes)

AC Termination Model Country-based Impedance-based Auto-Detected

Country-based

Impedance-based

Number of Rings: (1-50. Default 4)
(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

PSTN Ring Thru FXS: No Yes (Default Yes)
(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay (sec): (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): (2-10 seconds. Default 6 seconds)
(Used to detect PSTN hangup when FXO port is not answered)

PSTN Idle Wait Timeout between Outgoing Calls: (0-10 seconds. Default 4 seconds)

Figure 8: Configure FXO Port on the HT813 - FXO Termination

- First, we should confirm which method the PSTN line is using.

If the PSTN line is using current disconnect (typical case in North America), then we should turn on "Enable Current Disconnect" and disable "Enable PSTN Disconnect Tone Detection".

The default "Current Disconnect Threshold" is 100ms, but if you start experiencing dropped calls then you should raise this value by 100ms intervals.

If the PSTN disconnects using the tones method, then turn on "Enable PSTN Disconnect Tone Detection" and turn off the "Enable Current Disconnect" option.

For PSTN tone detection, the tone disconnect method is widely used everywhere else in the world. The North American busy tone value is "f1=480@-32,f2=620@-32,c=500/500" but these tones vary from country to country. You may look up for the settings for your country at www.3amsystems.com or download the information from <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>.

- Set "Number of Rings" option to 1. If you happen to experience caller ID issue, you may set it to 2 or 4.
- Set "PSTN Ring Thru FXS" to "No" if you prefer not to ring the FXS port on incoming PSTN calls after the Ring Thru Delay. In the sample setup, it's set to "Yes".



- Set "PSTN Ring Thru Delay" option to 1. If you happen to experience caller ID issue, you may set it to 2.
- Set the "Stage Method (1/2)" to 2 for 2-stage dialing.

<i>Stage Method (1/2):</i> <input type="text" value="2"/> (Default 2 - 2 stage dialing)

Figure 9: Configure FXO Port on the HT813 - Channel Dialing

Configure Unconditional Call Forward on HT813

On the HT813 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the IVR extension on the FCM630A. In this example, the FCM630A IP address is 192.168.5.190.

	User ID	Sip Server	Sip Destination Port
<i>Unconditional Call Forward to VOIP:</i>	<input type="text" value="7000"/>	@ <input type="text" value="192.168.5.190"/>	: <input type="text" value="5060"/>

Figure 10: HT813 Basic Settings

How to Dial

Once the HT813 and the FCM630A are set up as above, the inbound call and the outbound call will be working as described below.

- **Outbound call**
The extension registered to the FCM630A can dial the HT813's FXO extension number (1001 in this example). After you get the second dial tone, you can then dial a PSTN network number. Basically, the outbound call is done in a 2-stage manner.
- **Inbound call**
The user from outside network can dial into the PSTN line's number (connected to HT813). And then he/she will reach the IVR of the FCM630A. The IVR on FCM630A would allow the user to further enter extension number or key pressing digit to reach the desired destination.



METHOD 2: CONNECT FCM630A TO HT813 USING PEER SIP TRUNK

Create IVR on FCM630A

On the FCM630A web GUI, create an IVR extension under **Call Features**→**IVR**.

In IVR settings, if "Dial Other Extensions" is enabled, the calls dialing into the FCM630A IVR will be able to reach the internal extensions registered to the FCM630A. Also, you can assign the "Key Pressing Event" to different destinations.

The screenshot displays the 'Create New IVR' configuration page. The 'Basic Settings' tab is active. The 'Dial Other Extensions' section is highlighted with a red box, showing the 'Extension' radio button selected. Other settings include Name: HT813_IVR, Extension: 7001, IVR Black/Whitelist: Disable, and various timeout and prompt options.

* Name:	HT813_IVR
* Extension:	7001
Dial Trunk:	<input type="checkbox"/>
Dial Other Extensions:	<input type="checkbox"/> All <input checked="" type="checkbox"/> Extension <input type="checkbox"/> Conference <input type="checkbox"/> Video Conference
	<input type="checkbox"/> Call Queue <input type="checkbox"/> Ring Group <input type="checkbox"/> Paging/Intercom Groups
	<input type="checkbox"/> Voicemail Groups <input type="checkbox"/> Fax Extension <input type="checkbox"/> Dial By Name
* IVR Black/Whitelist:	Disable
Replace Display Name:	<input type="checkbox"/>
Return to IVR Menu:	<input type="checkbox"/>
Alert-info:	None
* Prompt:	welcome Upload Audio File
	Add Prompt +
* Digit Timeout:	3
* Response Timeout:	10
* Response Timeout Prompt:	ivr-create-timeout Upload Audio File
* Invalid Input Prompt:	invalid Upload Audio File
* Response Timeout Prompt Repeats:	3
* Invalid Input Prompt Repeats:	3
Language:	Default

Figure 11: Create IVR 7000 on the FCM630A

Create Peer SIP Trunk on FCM630A

On the FCM630A web GUI, create a peer SIP trunk under **Extension/Trunk**→**VoIP Trunks**. In this example, the HT813 IP address is 192.168.5.144.



Create New SIP Trunk

Type: Peer SIP Trunk

* Provider Name: HT813

* Host Name: 192.168.5.144

Keep Original CID:

Keep Trunk CID:

NAT:

Disable This Trunk:

TEL URI: Disabled

Caller ID:

CallerID Name:

Auto Record:

Direct Callback:

Cancel Save

Figure 12: Create Peer SIP Trunk on the FCM630A

Configure Outbound Rule on FCM630A

On the FCM630A web GUI, go to **Extension/Trunk**→**Outbound Routes** to create a new outbound rule. This would allow the extension on the FCM630A to reach numbers in PSTN network via the peer SIP trunk we just configured.



Figure 13: Configure Outbound Rule on the FCM630A

In this example "9x.", 9 is the first dialing digit and it will be stripped off when the call goes out.

Configure Inbound Rule on FCM630A

On the FCM630A web GUI, go to **Extension/Trunk**→**Inbound Routes** to create a new inbound rule. In this example, we create the DID as **20000**, which will be used in the HT813 call forward setting.



Figure 14: Configure Inbound Rule on the FCM630A

The default destination is configured to IVR.

Configure FXO Port on HT813

1. Connect the PSTN line to the HT813 FXO port.
2. On the HT813 web GUI, go to FXO Port setting page, configure the FXO port to send signaling SIP messages to the FCM630A's IP address. Please refer to the highlighted settings and other necessary settings in the following figures.

You can set anything you want on the SIP user ID, authentication ID and username. We choose 1111 in our example.

In this example, the FCM630A IP address is 192.168.5.190.



Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT	FXO PORT
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Account Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: No Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

SIP Transport: UDP TCP TLS (default is UDP)

NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV

DNS SRV use Registered IP: No Yes

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 15: Configure FXO Port on the HT813 - Registration

Since we are going to use IVR when the call is forwarded to the FCM630A, the FCM630A will need to be able to detect the DTMF digits. Configure the HT813 FXO port DTMF settings as below for an initial setup.

<i>Preferred DTMF method</i>	Priority 1:	<input type="text" value="RFC2833"/>
<i>(in listed order):</i>	Priority 2:	<input type="text" value="SIP INFO"/>
	Priority 3:	<input type="text" value="In-audio"/>

Figure 16: Configure FXO Port on the HT813 - DTMF Settings

There are a few necessary changes to be made in FXO termination section and Channel Dialing section.



FXO Termination

Enable Current Disconnect: No Yes (Default Yes. If set to yes, enter threshold below)

Current Disconnect Threshold (ms): (50-800 milliseconds. Default 100 milliseconds)

Enable PSTN Disconnect Tone Detection: No Yes (Default No)
(If set to yes, the following tone is used as the disconnect signal)

PSTN Disconnect Tone: (Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)
(Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)
(Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)

Enable Polarity Reversal: No Yes (Default No. Check with your PSTN carrier before setting to Yes)

AC Termination Model Country-based Impedance-based Auto-Detected

Country-based

Impedance-based

Number of Rings: (1-50. Default 4)
(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

PSTN Ring Thru FXS: No Yes (Default Yes)
(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay (sec): (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): (2-10 seconds. Default 6 seconds)
(Used to detect PSTN hangup when FXO port is not answered)

Figure 17: Configure FXO Port on the HT813: FXO Termination

- First, we should confirm which method the PSTN line is using.

If the PSTN line is using current disconnect (typical case in North America), then we should turn on "Enable Current Disconnect" and disable "Enable PSTN Disconnect Tone Detection".

The default "Current Disconnect Threshold" is 100ms, but if you start experiencing dropped calls then you should raise this value by 100ms intervals.

If the PSTN disconnects using the tones method, then turn on "Enable PSTN Disconnect Tone Detection" and turn off the "Enable Current Disconnect" option.

For PSTN tone detection, the tone disconnect method is widely used everywhere else in the world. The North American busy tone value is "f1=480@-32,f2=620@-32,c=500/500" but these tones vary from country to country. You may look up for the settings for your country at www.3amsystems.com or download the information from <http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>.

- Set "Number of Rings" option to 1. If you happen to experience caller ID issue, you may set it to 2. In the sample setup, it's set to 2.
- Set "PSTN Ring Thru FXS" to "No".



- Set "PSTN Ring Thru Delay" option to 1. If you happen to experience caller ID issue, you may set it to **2. In the sample setup, it's set to 2.**
- Set the "Wait for Dial-Tone" to "No".
- Set the "Stage Method (1/2)" to 1.

Wait for Dial-Tone: <input checked="" type="radio"/> No <input type="radio"/> Yes (Default Yes - dial upon dial-tone) Stage Method (1/2): <input type="text" value="1"/> (Default 2 - 2 stage dialing)

Figure 18: Configure FXO Port on the HT813 - Channel Dialing

Exchange SIP Port Settings for FXS and FXO on HT813

- On the HT813 web GUI, go to FXO setting page, configure the "Local SIP Port" to be 5060. (The default setting is 5062.)
- On the HT813 web GUI, go to FXS setting page, configure the "Local SIP Port" to be 5062. (The default setting is 5060.)

Configure Unconditional Call Forward on HT813

On the HT813 web GUI, go to Basic setting page, configure "Unconditional Call Forward to VOIP" to the DID number **20000**. This is the same number configured in FCM630A inbound route dial pattern. In this example, the FCM630A IP address is 192.168.5.250.

Unconditional Call Forward to VOIP:	User ID <input type="text" value="20000"/>	Sip Server <input type="text" value="@192.168.5.190"/>	Sip Destination Port <input type="text" value="5060"/>
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Figure 19: HT813 Basic Settings

How to Dial

Once the HT813 and the FCM630A are set up as above, the inbound call and the outbound call will be working as described below.

- **Outbound call**
The extension registered to the FCM630A can dial prefix + PSTN number to reach outside numbers in PSTN network, as defined in FCM630A outbound route.

- **Inbound call**

The user from outside network can dial into the PSTN line's number (connected to HT813). And then he/she will reach the IVR of the FCM630A. The IVR on FCM630A would allow the user to further enter extension number or key pressing digit to reach the desired destination. The inbound call will go through the inbound route set up on the FCM630A.

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