

FIBERME Communications LLC.

Configuring FCM630A Series with HT813

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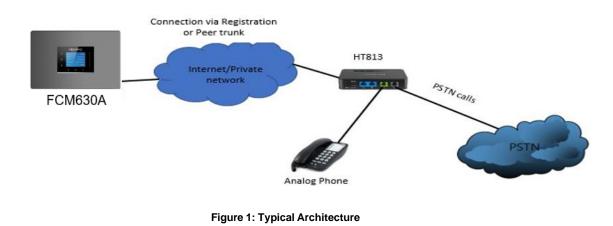
OVERVIEW

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:



▲ _{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.

Create New Extension						
Basic Settings Media	Features	Specific Time	Follow Me			Cancel Save
* Select Extension Type :	SIP Extension		~			
Select Add Method:	Single		~			
General						
* Extension :	1000			CallerID Number:		
* Permission :	Internal		~	* SIP/IAX Password :	d~HBGjj7	
AuthID :				Voicemail:	Local Voicemail	~
* Voicemail Password :	1413097			Skip Voicemail Password		
				Verification:		
Send Voicemail to Email :	Default		~	Keep Voicemail after	Default	~
				Emailing:		
Enable Keep-alive :				* Keep-alive Frequency :	60	

Figure 2: Create Extension 1000 on the FCM630A

Create New Ex	tension							
Basic Settings	Media	Features	Specific Time	Follow Me			Cancel	Save
* Select Extension	Type:	SIP Extension		~				
Select Add Meth	nod:	Single		~				
General								
* Extension :		1001			CallerID Number:			
* Permission :		Internal		~	* SIP/IAX Password :	ru^h8ByT		
AuthID :					Voicemail:	Local Voicemail	~	
* Voicemail Pass	sword :	5615081			Skip Voicemail Password			
					Verification:			
Send Voicemai	il to Email :	Default		~	Keep Voicemail after	Default	~	
					Emailing:			
Enable Keep-a	live :				* Keep-alive Frequency :			

Figure 3: Create Extension 1001 on the FCM630A



Create IVR on FCM630A

On the FCM630A web GUI, create an IVR extension under **Call Features** \rightarrow **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 ableto reach the internal extensions registered to the FCM630A. Also, you can assign the "Key Pressing Event" to different destinations.

Create New IVR			
Basic Settings Key Press	ing Events		Cancel Save
* Name :	HT813_IVR]	
* Extension :	7001	- -	
Dial Trunk :			
Dial Other Extensions :	All ☑ Extension Conference Vid Call Queue Ring Group Paging/Intri Voicemail Groups Fax Extension D	ercom Groups	
* IVR Black/Whitelist:	Disable ~		
Replace Display Name :			
Return to IVR Menu :			
Alert-info :	None v		
* Prompt:	welcome ×	Upload Audio File	
	Add Prompt	0	
* Digit Timeout :	3		
* Response Timeout:	10		
* Response Timeout Prompt:	ivr-create-timeout v	G Upload Audio File	
* Invalid Input Prompt :	invalid v	Upload Audio File	
* Response Timeout Prompt Repeats :	3 ~		
* Invalid Input Prompt Repeats :	3 ~		
Language:	Default		

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				
I	Primary SIP Server:	192.168.5	.190	1	(e.g., sip.mycompany.com, or IP address)		
1	Failover SIP Server:				(Optional, used when primary server no		
		response)					
Prefer I	Primary SIP Server:	No	Yes	(yes - will reg	rister to Primary Server if Failover registration		
		expires)			(e.g., proxy.myprovider.com, or IP address, if		
	Outbound Proxy:	any)			(e.g., proxy.myprovider.com, of 1P address, if		
P 1					(e.g., proxy.myprovider.com, or IP address, if		
Backu	p Outbound Proxy:	any)					
Prefer Prima	ry Outbound Proxy:	No	Yes	(yes - will rer	egister via Primary Outbound Proxy if registration		
		expires)					
Allow DHCP Optic	on 120 (override SIP server):	No	Yes				
	SIP Transport:	• UDP	O TC	P 🔍 TLS	(default is UDP)		
SIP URI Schem	e When Using TLS:	◎ sip	sips				
	eral Port in Contact	No	• Yes				
	with TCP/TLS:	• No	Yes				
	NAT Traversal:	No	Keep	-Alive 🔍 ST	UN OUPnP		
	SIP User ID:	1000			(the user part of an SIP address)		
	Authenticate ID:	1000			(can be identical to or different from SIP User		
1.00		ID)					
Aut	henticate Password:				(purposely not displayed for security protection)		
	Name:				(optional, e.g., John Doe)		
Date of	DNS Mode:	A Rec		SRV ONA	APTR/SRV		
DNS SI	RV use Registered IP:	• No	O Yes				
	Tel URI:	Disabled	•	1			
7	SIP Registration:	O No	• Yes	1			
	Inregister On Reboot:	No	Ves				
Outgoing Call	without Registration:	No	O Yes		T040		

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.



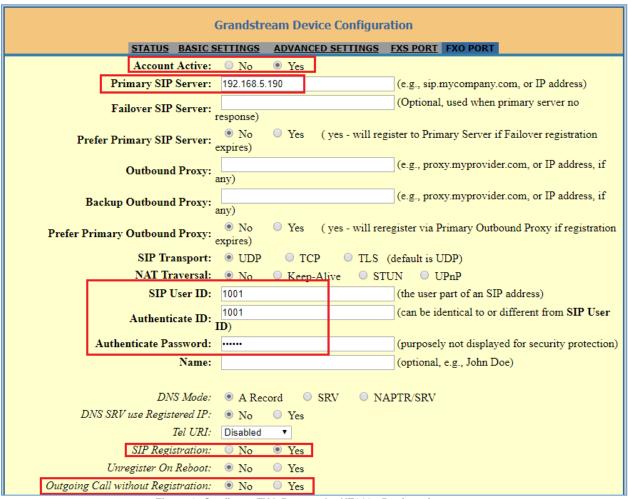


Figure 6: Configure FXO Port on the HT813 - Registration

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.

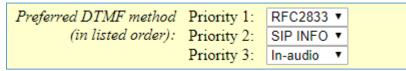


Figure 7: Configure FXO Port on the HT813 - DTMF Settings

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):	100 (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:	f1=480@-32,f2=620@-32,c=500/500;
	(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 O Impedance-based O Auto-Detected
Country-based	USA 🔻
Impedance-based	600R 600 ohms
Number of Rings:	1 (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 Ves (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 (1-10 seconds. Default 4 seconds)
PSTN Ring Timeout (sec):	6 (2-10 seconds. Default 6 seconds)
	(Used to detect PSTN hangup when FXO port is not answered)
PSTN Idle Wait Timeout	4 (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. TheNorth 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 <u>www.3amsystems.com</u> or download the information from <u>http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf</u>.

- 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.

Stage Method (1/2):	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 I	D Sip Server	Sip Destination Port
Unconditional Call Forward to VOIP: 7000	@ 192.168.5.190	: 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.

Create New IVR			
Basic Settings Key Pressi	ng Events		Cancel Save
			*
* Name :	HT813_IVR		
* Extension :	7001		
Dial Trunk:			
Dial Other Extensions :	□ All		
	Call Queue Ring Group Paging/Inter		
	Voicemail Groups Fax Extension Dia	il By Name	
* IVR Black/Whitelist:	Disable v		
Replace Display Name :			
Return to IVR Menu:			
Alert-info :	None ~		
* Prompt:	welcome ~	🔥 Upload Audio File	
	Add Prompt	0	
* Digit Timeout :	3		
* Response Timeout:	10		
* Response Timeout Prompt :	ivr-create-timeout v	🔥 Upload Audio File	
* Invalid Input Prompt :	invalid v	🔥 Upload Audio File	
* Response Timeout Prompt	3 ~		
Repeats :			
* 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:	

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.



Create New Outbound Rul	e				Cancel	Save
General						^
* Calling Rule Name :	HT813_Outbound		Disable This Route:			
* Pattern :	_9x.		Privilege Level:	Internal	·	
		li.		Warning: Setting privilege level at "Internal" has potential security risks.		
PIN Groups:	None	~	PIN Groups with Privilege			
Password :			Level:			
Enable Filter on Source Caller	D					
Enable Filter on Source Caller			Outbound Route CID:			
Call Duration Limit						
Main Trunk						
*Trunk:	SIPTrunks HT813	~				
Strip:	1					
Prepend :						

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.



Create New Inbound Rule	<u>,</u>			Cancel Save
* Trunks:	SIPTrunks HT813	~		
* Pattern :	_200000		CallerID Pattern :	
				1
Disable This Route:			Allowed to seamless	
			transfer:	
Alert-info:	None	~		
Fax Detection :				
Block Collect Calls:			Prepend Trunk Name:	
Set CallerID Info :			Enable Route-Level Inbound	
			Mode:	
Inbound Multiple Mode:				
Default Mode Mode 1				
* Default Destination :	IVR.	V HT813_IVR	×	
Time Condition				
Add				

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 S	ETTINGS ADV	ANCED SETTINGS	FXS PORT FXO PORT	
	Account Active:	🔍 No 🔍 Y	es		
	Primary SIP Server:	192.168.5.190		(e.g., sip.mycompany.com, or IP address)	
	Failover SIP Server:			(Optional, used when primary server no	
	1	response)			
Р	refer Primary SIP Server:	 No Yes (yes - will register to Primary Server if Failover registration expires) 			
				(e.g., proxy.myprovider.com, or IP address, if	
	Outbound Proxy:	(e.g., prospin) providencem, or in address, in my)			
	Backup Outbound Proxy:			(e.g., proxy.myprovider.com, or IP address, if	
	any)				
Prefer 1	Primary Outbound Proxy:		es (yes - will rere	gister via Primary Outbound Proxy if registration	
	SIP Transport:	expires)	TCP OTLS (default is UDP)	
	NAT Traversal:		leep-Alive OSTU	· · · · · · · · · · · · · · · · · · ·	
	SIP User ID:	1111	eep-Anve 0 510	(the user part of an SIP address)	
	SH CSH ID.	1111		(can be identical to or different from SIP User	
	Authenticate ID:			(can be identical to of different from 511 Cser	
	Authenticate Password:			(purposely not displayed for security protection)	
	Name:	1111		(optional, e.g., John Doe)	
DNS Mode: A Record SRV NAPTR/SRV 					
i	DNS SRV use Registered IP: 💿 No 💿 Yes				
Tel URI: Disabled					
	SIP Registration: No Yes 				
	Unregister On Reboot: 💿 No 🛛 🔍 Yes				
Outgoin	g Call without Registration:	🔍 No 🔍 Y	es		

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.

Preferred DTMF method	Priority 1:	RFC2833 •	
(in listed order):	Priority 2:	SIP INFO 🔻	
	Priority 3:	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):	100 (50-800 milliseconds. Default 100 milliseconds)			
Enable PSTN Disconnect Tone Detection:	No O Yes (Default No)			
	(If set to yes, the following tone is used as the disconnect signal)			
PSTN Disconnect Tone:	f1=480@-32,f2=620@-32,c=500/500;			
	(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 Ves (Default No. Check with your PSTN carrier before setting to Yes)			
AC Termination Model	Country-based Impedance-based Auto-Detected			
Country-based	USA			
Impedance-based	600R 600 ohms			
Number of Rings:	2 (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 O 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):	2 (1-10 seconds. Default 4 seconds)			
PSTN Ring Timeout (sec):	6 (2-10 seconds. Default 6 seconds)			
	(Used to detect PSTN hangup when FXO port is not answered)			
Eigure 17	Configure EXO Port on the HT813: EXO 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 thenyou 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. TheNorth 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 <u>www.3amsystems.com</u> or download the information from <u>http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf</u>.

- 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:
 Image: No
 Yes
 (Default Yes - dial upon dial-tone)

 Stage Method (1/2):
 1
 (Default 2 - 2 stage dialing)

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

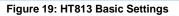
Exchange SIP Port Settings for FXS and FXO on HT813

- 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.

User ID	Sip Server	Sip Destination Port
Unconditional Call Forward to VOIP:	@ 192.168.5.190	: 5060



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 enterextension 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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