

Grandstream Networks, Inc.

Connecting UCM6XXX with HT841/HT881 FXO Gateway Configuration Guide



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Introduction

This document describes basic configuration to interconnect UCM6XXX series and HT841/HT881. In this document, we are using HT841 as an example. The following steps can be used for the HT881 model as well. This is typically applied to the scenario where users would like to add a HT841/HT881 not only as a remote extension but also as an external PSTN trunk.



HT8x1 and UCM63xx Interconnection

There are two ways to set up the UCM63XX series IP PBX with the HT841/HT881.

- Method 1: Configure HT841/HT881 as a SIP Peer Trunk.
- Method 2: Register HT841/HT881 on the UCM6XXX directly as an extension.

Method 1: Connect UCM6XXX to HT841 using Peer SIP Trunk

Create Peer SIP Trunk on UCM6XXX

On the UCM6XXX web GUI, create a peer SIP trunk under **Extension/Trunk** ->**VOIP Trunks**. In this example, the HT841/HT881 IP address is 192.168.5.79, followed by 6060 which is the default listening port of FXO 1, on which the analog line is connected.

S UCM6302			
	VoIP Trunks > Create New	SIP Trunk	
🝰 Extension/Trunk 🔨	If the host is not a numeric IP addre	ss, but the port number is present in the URI, the UCM p	erforms an A or AAAA record lo
Extensions	Disable This Trunk		
Extension Groups	Туре	Peer SIP Trunk	
Analog Trunks	= Provider Name	HT841	
VolP Trunks	Host Name	192.168.5.79:6060	
WebRTC Trunks	Transport	UDP	· ·
SLA Station	Keep Original CID		
Outbound Routes	Keep Trunk CID		
InDound Routes	NAT		
E Maccasing	TEL URI	Disabled	
RR PBX Settings	CallerID Number		
System Settings	Callerin Name		
A Contacts	California Marrie		
Device Management ~	Auto Record		
₩ Maintenance ✓	Direct Callback		
È CDR ✓		Cancer Save	

Create Peer SIP Trunk on the UCM6XXX

Configure Outbound Rule on UCM6XXX

On the UCM6XXX web GUI, go to **Extension/Trunk** ->**Outbound Routes** to create a new outbound rule. This would allow the extension on the UCM6XXX to reach numbers in PSTN network via the peer SIP trunk we just configured.

Outbound Routes > Edit Outbound R	tule: HT841_Outbound				
General					
* Outbound Rule Name	HT841_Outbound		Disable This Route		
* Pattern	_9x.		Privilege Level	Internal	u.
				Warning: Setting privilege level at "Internal" has potential security risks.	
Path Courses	l line l	1	PNI Courses with Privilege Level		
Pin Groups	NOTE -		Pin aroups with Privilege Level		
Password			Auto Record		
Local Country Code					
C Enable Source Caller ID Whitelist					
Enable Source Caller ID Whitelist			Outbound Route CID		
Call Duration Limit					
Call Duration Limit					
Main Trunk					
* Trunk	SIP Trunks HT841 v				
Strip	1				

Configure Outbound Rule on the UCM6XXX

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

Configure Inbound Rule On UCM6XXX

On the UCM6XXX web GUI, go to Extension/Trunk ->Inbound Rules to create a new inbound rule.

In this example, we create the DID as 15555551234, this will be used in the HT841/HT881 call forward setting.

Inbound Routes > Create New	Inbound Rule		
General			
• Trunks	SIP Trunks HTB41	Inbound Route Name	
Disable This Route			
Pattern			
Pattern	_15555551234	CallerID Pattern	
	0.		
CID Source	None	Seamless Transfer Whitelist	
Call Setting			
Ringback Tone	None	Alert-info	None
Auto Record		Fax Detection	
Block Collect Calls			
CallerID Setting			
	Cancel Save		

Configure Inbound Rule on UCM6XXX

The default destination is configured to IVR.

Default Mode 1							
Default Destination	IVR		~ HT8	41_IVR	v		
Time Condition							
Add							
Time Condition	Time	Week	Month	Day	Destination	Options	

Configure Inbound Default Mode

Configure FXO Port on HT841 when Peered with UCM6XXX

- Connect the PSTN line to the HT841 FXO port.
- On the HT841 web GUI, go to the **FXO Profile** settings page and enter the IP address of the UCM6XXX that you are peering with. In the following example, UCM6XXX has IP address 192.168.5.105

FXO PROFILE 1			
General Settings SIP Settings C	Codec Settings Call Se	ettings FXO Termination Channel Dialing	
Account Registration			
_	Profile Active 🕐	🔿 No 💿 Yes	
[Primary SIP Server 🕚	192.168.5.105	
	Failover SIP Server 🕚		
Prefe	er Primary SIP Server 🕚	No	
	Outbound Proxy 🕐		
Bac	sup Outbound Proxy 🛞		
Prefer Prim	ary Outbound Proxy 🕚	No Yes	
	From Domain 🕚		

 $\circ~$ Please make sure the $\ensuremath{\text{SIP Registration}}$ option under is set to $\ensuremath{\text{No}}.$

O PROFILE 1	
neral Settings SIP Settings Codec Settings	Call Settings FXO Termination Channel Dialing
IP Basic Settings	
SIP Registra	ation 🕐 💿 No 🔅 Yes
SIP Trans	iport 🕐 💿 UDP 🔿 TLS
Unregister On Rei	boot 🕐 💿 No 💿 All 💿 Instance
Outgoing Call without Registre	ation 🕐 🔿 No 🐵 Yes
Register Expire	ation 🕚 60
Reregister before Expira	ation 🕐 0

• There are few changes to be made in FXO termination section.

FXO PROFIL	E 1				
General Settings	SIP Settings	Codec Settings	Call Settings	FXO Termination	Channel Dialing
	Er	able Current Disconn	ect 🕐 💿 No	🔿 Yes	
	Current Di	sconnect Threshold (r	ms) 🕐 100		
	Enable PSTN Di	sconnect Tone Detect	ion 🕑 🕕 No	Yes	
		Enable Polarity Rever	rsal 🕐 💿 No	O Yes	

- First we should confirm which method the PSTN line is using.
 - 1. If your PSTN line uses current disconnect (common in North America), enable 'Current Disconnect' and disable 'PSTN Disconnect Tone Detection.' The default 'Current Disconnect Threshold' is 100ms; increase it in 100ms increments if call drops occur.
 - 2. 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 http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
- Configure the Call Progress tones under **System Settings => Ringtone**, you can also keep them to the Default value as shown in the screenshot below:

Ringtone		
System Ring Cadence 💿	c=2000/4000;	
Prompt Tone Access Code ()		
PSTN Disconnect Tone 🕚	fxoch1-4.f1=480@-32,f2=620@-32,c=500/500;	
CPT Settings		
Dial Tone 🕚	fxsch1:f1=350@-17,f2=440@-17,c=0/0;fxoch1-4:f1=35(
Ringback Tone 🕚	fxsch1:f1=440@-17,f2=480@-17,c=2000/4000.fxach1-4	
Busy Tone 🕚	fxsch1:f1=480@-21,f2=620@-21,c=500/500;fxoch1-4.f	
Rearder Tone 🕐	fxsch1;f1=480@-21,f2=620@-21,c=250/250;fxoch1-4;f	
Confirmation Tone 💿	fxsch1:f1=350@-11,f2=440@-11,c=100/100-100/100-1(
Call Waiting Tone 🕚	frsch1:f1=440@-13,c=300/10000;fxoch1-4;f1=440@-11	
Wait for Dial-Tone 🕚	fxsch1:f1=350@-17,f2=440@-17,c=0/0;fxoch1-4:f1=35(
Conference Party Hangup Tone 🕚	fxsch1:f1=425@-15,c=600/600;fxoch1-4:f1=425@-15,c	
Special Proceed Indication Tone	fxsch1;f1=350@-13;f2=440@-13;c=750/750;fxoch1-4;f	
Special Condition Tone	fxsch1;f1=350@-13;f2=450@-13;c=750/750;fxoch1-4;f	
	Syntax: fxsch1:f1=val[,f2=val[,c=on1/off1[-on2/off2[-on	3/off3]]]]; fxoch x-y:f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]];

Configure FXO Port on the HT841: Call Progress Tones

• Set "Number of Rings" option to 4, it will ring 4 times before sending the call through the SIP trunk.

Enable PSTN Disconnect Tone Detection 3	No Yes	
Enable Polarity Reversal	No Ves	
Polarity On Answer Delay 🔅	1000	
AC Termination Model 🔮	Country-based Impedance-based Auto-Det	etected ()
Country-based @	USA 🗸 🗸	
Impedance-based	600R - 600 ohms 🗸 🕚	
Number of Rings ③	4	
PSTN Ring Thru FXS (No Yes	

Configure FXO Port on the HT841: Number of Rings

• Set the "PSTN Ring Thru FXS" option to "No", that way the call is routed through the defined FXO

	Number of Rings 💿	4	
	PSTN Ring Thru FXS 🕚	No ○ Yes	
PSTN R	ing Thru Delay (sec) 💿	4	
PSTN	N Ring Timeout (sec) 💿	6	
PSTN Idle Wait Timeout betw	veen Outgoing Calls 🕚	4	
PIN for	r VolP-to-PSTN Calls 🕚		¥
PIN for	r PSTN-to-VoIP Calls 🕚		~
Enable Early Media for	r VolP-to-PSTN Calls 🕚	No Yes	

Configure PSTN Ring Thru FXS

• Under Channel Dialing :

1. Set the "Wait for Dial-Tone" to "No".

2. Set the "Stage Method (1/2)" to 1.

XO PROFILE	1				
General Settings	SIP Settings	Codec Settings Call Se	ttings FXO Terminatio	n Channel Dialing	
		DTMF Digit Length (ms) 🕚	100		
		DTMF Dial Pause (ms) 🕚	100		
		First Digit Timeout (sec) 🕐	10		
		Inter-Digit Timeout (sec) 🔘	4		
		Wait for Dial-Tone	⊙ No 🔿 Yes		
		Stage Method (1/2)	1		
	Min Delay f	Before Dial PSTN Number	500		
			Save	Save and Apply	

Configure FXO Port on the HT841: Channel Dialing

Note

• The formula of defining the FXO SIP ports is the following: SIP port = Profile's base SIP port + (Port id - 1) *2, while the Port Id is defined as follows: FXS = 0, FXO1-8 = 1 - 8 for HT881 and FXO1-4 = 1 - 4 for HT841, in this case, if we set FXO profile 1 local SIP port to 6060, then FXO1 will use port 6060, FXO2 will use 6062, FXO3 will use 6064 and so on...

Configuring Unconditional Call Forward on HT841

On the HT841 web GUI, go to the **FXO Ports** page, configure "Unconditional Call Forward to VOIP" to the DID number 15555551234, this is our TEL-to-IP call.

In this example, we will use the SIP server for FXO Profile1, which is the IP Address of the UCM6xxx, and for the port number, we set it to 5060.

Uncondition	onal Call Forward to VOIP				
FXO PORTS	CID		Sip Server	5	ip Destination Port
1	15555551234	@	192.168.5.105		5060
2		0			
3		@			
4		0			

HT841- Call Forwarding

Method 2: Register HT841 on UCM6XXX as an Extension

Create SIP Extension on UCM6XXX

On the UCM6XXX web GUI, create an extension under **Extension/Trunk→Extensions**. This extension is used for HT841 FXO registration.

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

Extensions > Edit Extension: 1001							
Basic Settings Media Featu	res Voicemail	Specific Time	Wave Client	Follow Me	Advanced Settings		
General							
Extension					CalleriD Number		
Call Privileges	Local		~		 SIP/IAX Password 		
AuthID					 Concurrent Registrations 	3	
User Settings							
First Name					Last Name		
Email Address					* User/Wave Password	*****	
User Portal/Wave Privileges	Default		~		Mobile Number	+1 ~	
	Add / Edit Privileges						
Department					Job Title		
Contact Privileges							
Same as Department Contact Privileges					* Contact View Privileges	All Contacts	
						Add / Edit Privileges	
Sync Contact							
	Cancel Save						

Create SIP Extension on UCM6XXX For FXO port

Configure HT841 User Setting as an Extension Registered on UCM6XXX

Under HT841 web GUI->**Ports**, please enter the SIP Extension information created earlier in the UCM6XXX for the FXO port, In this example, extension 1001 is used in order to register HT841' FXO port as an extension user on UCM6XXX.

PORTS	SIP User ID	Authenticate ID	Authenticate Password	Name	Profile ID	Hunting Group	Request URI Routing ID	Enab	le Port	Unconditional Call Forward to PSTN
1	1001	1001		1001	FXO PROFILE 2	Disabled ~		O No	Yes	
2			<u> </u>		FX0 PROFILE 1	Disabled V		O No	Yes	
3			54		FXO PROFILE 1	Disabled		O No	Yes	
4			~		FXO PROFILE 1	Disabled V		O No	Yes	

Registering extension on HT841

Under HT841 web GUI, FXO Profile 2, please fill in UCM6XXX information as explained in method 1.

FXO PROFILE 2					
General Settings SIP Settings Codec Settings Call Se	ttings FXO Termination Channel Dialing				
Account Registration					
Profile Active 💿	🔿 No 💿 Yes				
Primary SIP Server	192.168.5.105				
Failover SIP Server 🕚					
Prefer Primary SIP Server 💿	No				
Outbound Proxy 🕚					
Backup Outbound Proxy 🕥					
Prefer Primary Outbound Proxy 🕥	No Yes				
From Domain 🕐					

Please make sure under **SIP Settings** tab, **SIP Registration** option is set to **Yes**, as it is required for HT841 to successfully register on UCM6XXX.

FXO PROFILE 2	
General Settings SIP Settings Codec Settings	Call Settings FXO Termination Channel Dialing
SIP Basic Settings	
SIP Registrat	on 🕚 🗌 No 💿 Yes
SIP Transp	ort 🕐 💿 UDP 🔿 TCP 🔅 TLS
Unregister On Reb	iot 💿 🖲 No 🗌 All 🔅 Instance
Outgoing Call without Registrat	on 🕲 🔿 No 💿 Yes
Register Expirat	on () 60
Reregister before Expirat	on 🕐 0
SIP Registration Failure Retry Wait Ti	ne 🕚 20
SIP Registration Failure Retry Wait Time upon 403 Forbide	en 🕐 1200

HT841 SIP Settings

We can check UCM6XXX SIP Extension Status to see if HT841 has been successfully registered as an extension device for the FXO port. The green icon indicates that HT841 is registered on UCM6XXX.

	• Idle	Available	1001	0/0/0	SIP(WebRTC)	192.168.5.106:7062	Synced	⊵₀	🗹 约 🖞 💼	

UCM6XXX – SIP Extension Status

Now HT841 is registered at UCM6XXX as an extension device. Please refer to method 1 in the previous section to adjust FXO Port and DTMF settings on HT841, Click **here** to access that section.

Configuring Unconditional Call Forward on HT841

On the HT841 web GUI, go to the **Ports** page, configure "Unconditional Call Forward to VOIP" to the extension number of the extension registered to the UCM63xx, this is our TEL-to-IP call.

On the HT841 web GUI, go to the FXO Ports, configure "Unconditional Call Forward to VOIP" to the IVR extension on the UCM6XXX. In this example, the UCM6XXX IP address is 192.168.5.105

Unconditional Call Forward to VOIP						
FXO PORTS	CID		Sip Server	5	ip Destination Por	
1	7000	@	192.168.5.105		5060	
2		@				
3		@				
4		@				



In order for this setup to work, it is extremely important that both the FXO gateway HT841/HT881 and the UCM63xx are located on the same LAN OR have Public Static IPs. In short, both devices should be able to locate each other.

How to dial

Once the HT841 and the UCM6XXX are set up as above, the inbound call and the outbound call will be working as described below.

• Outbound call

Method 1: The extension registered to the UCM6XXX can dial prefix + PSTN number to reach outside numbers in the PSTN network, as defined in the UCM6XXX outbound route.

Method 2: The extension registered to the UCM6XXX can dial HT841's FXO extension number (1001 in this example). After getting the second dial tone, you can then dial the PSTN network number. Basically, the outbound call is done in a 2-stage manner.

Example:

Method 1: The UCM-registered extension 1000 can call +1(555) 555-1234 by dialing 915555551234, with the '9' stripped and the rest of the number dialed.

Method 2: The UCM-registered extension 1000 can call the HT841 registered extension 1001, then dials the PSTN number +1(555) 555-1234

• Inbound call

The user from the outside network can dial into the PSTN line's number (connected to HT841/HT881). And then he/she will reach the IVR of the UCM6XXX. The IVR on UCM6XXX would allow the user to further enter the extension number or key pressing digit to reach the desired destination. The inbound call will go through the inbound route set up on the UCM6XXX.

Example:

A user dials the office number +1(555) 555-1234, this is the DID number provided by the PSTN service provider, then he will be prompted with an IVR, he can wait until he is prompted to call the desired extension, then he can call extension number 405 to reach his destination.

Supported Devices

Device	Firmware Required			
HT841	1057			
HT881	1.0.3.7+			
UCM6301/6302/6304/6308	1 0 22 17+			
UCM6300A/6302A/6304A/6308A	1.0.23.17+			
UCM6510	1.0.20.52+			
UCM6202/6204/6208	1.0.20.52+			

Supported Devices

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