Grandstream Networks, Inc.

How to Configure IPVideoTalk with UCM

Directory

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OVERVIEW

This document introduces the IPVideoTalk service configuration on Grandstream UCM device with IPVideoTalk server.

1. **Connect Grandstream UCM with IPVideoTalk conferencing system:**
   - Configure SIP Trunk server in IPVideoTalk server, which is the server address of Grandstream UCM.
   - Configure VoIP Trunk, Outbound Route and other information in Grandstream UCM.
2. **Calling out via UCM on IPVideoTalk server:**
   - Configure SIP Trunk server for calling out on IPVideoTalk server, which is the server address of Grandstream UCM.
   - Configure Inbound Route in Grandstream UCM.
3. **Introduce how to join into IPVideoTalk conferencing system for UCM users.**

IPVIDEOTALK SERVICE CONFIGURATION ON UCM

Configure SIP Trunk on IPVT

Log into IPVideoTalk portal with the enterprise account, and access to “Admin Center” -> “SIP Trunk Configuration” page. Fill the UCM server IP address (public IP address) in “Server IP address”, ~~blank, and~~ click to save the configuration as the figure shown below:

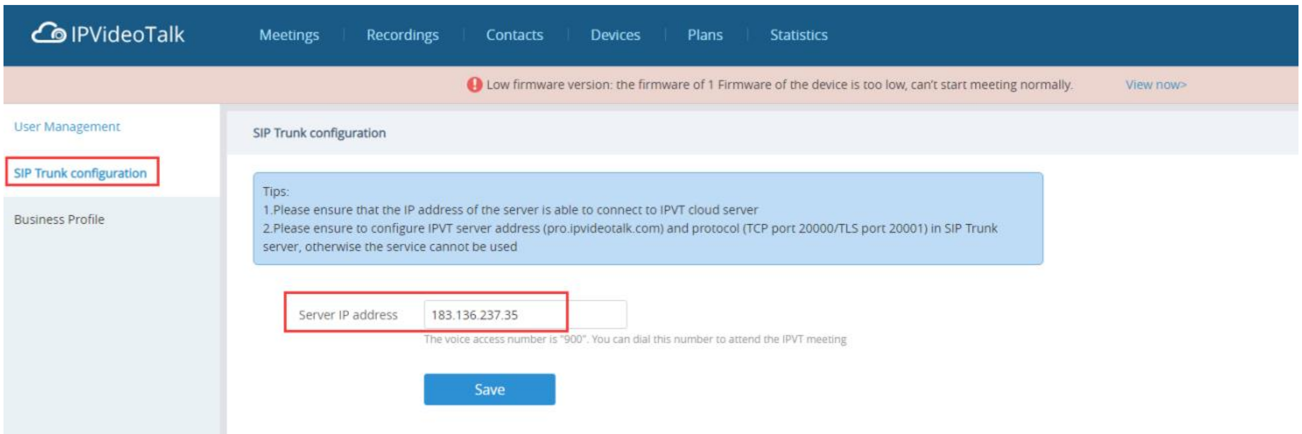

Figure 1: Configure IPVT SIP Trunk

Configure Grandstream UCM

Configure VoIP Trunk

1. Login Grandstream UCM Web UI, and access to “Extension / Trunk → VoIP Trunk”.

2. Select Create New SIP Trunk, and fill the information into option “Provider Name” and “Host Name” as the figure shows below:

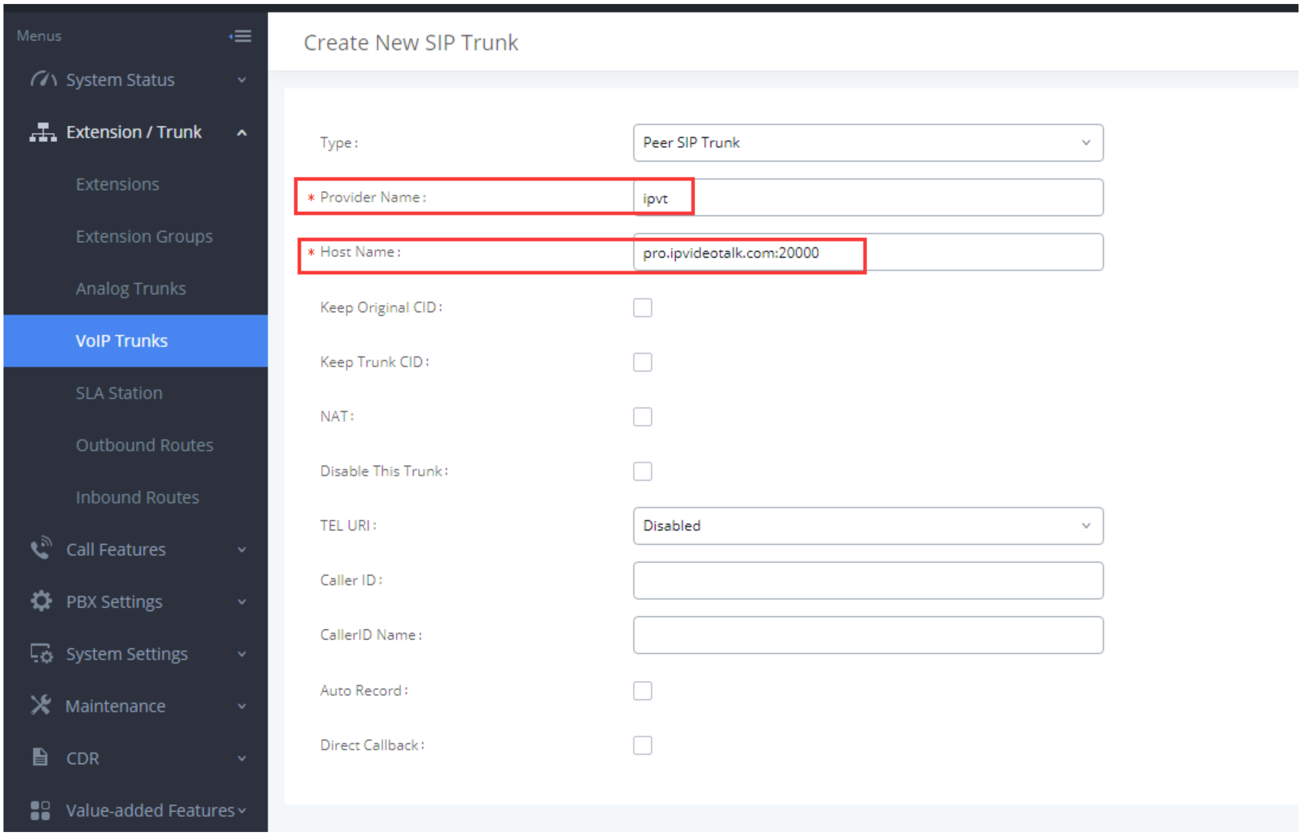

Figure 2: Configure VoIP Trunk

- **Provider Name:** Users need to fill in the provider name, and the duplicated name is not allowed. The provider name will be shown up during inbound/outbound routing.

Host Name: Fill in the IPVideoTalk server domain name and port. For different protocols, the port numbers are different.

TCP: pro.ipvideotalk.com:20000, TLS: pro.ipvideotalk.com:20001

3. Configure VoIP Trunk SIP Transport and Codecs

- **SIP Transport:** Users could select TCP/TLS as the SIP Transport, and the Port number should correspond the SIP Transport type. Please see the figure below:

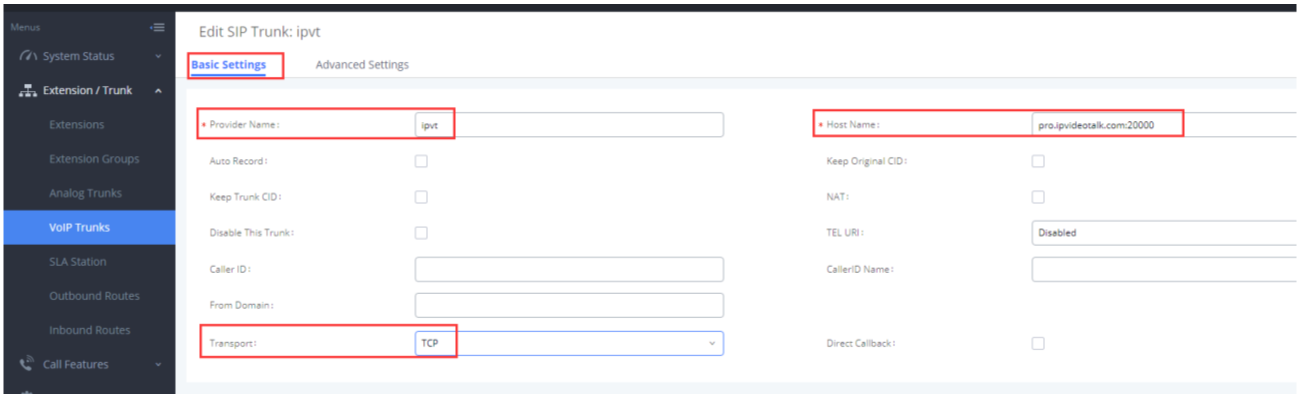

Figure 3: Configure SIP Transport

IPVT Mode: This option is necessary. If users do not check this option, the audio/video calls may generate abnormal problems.

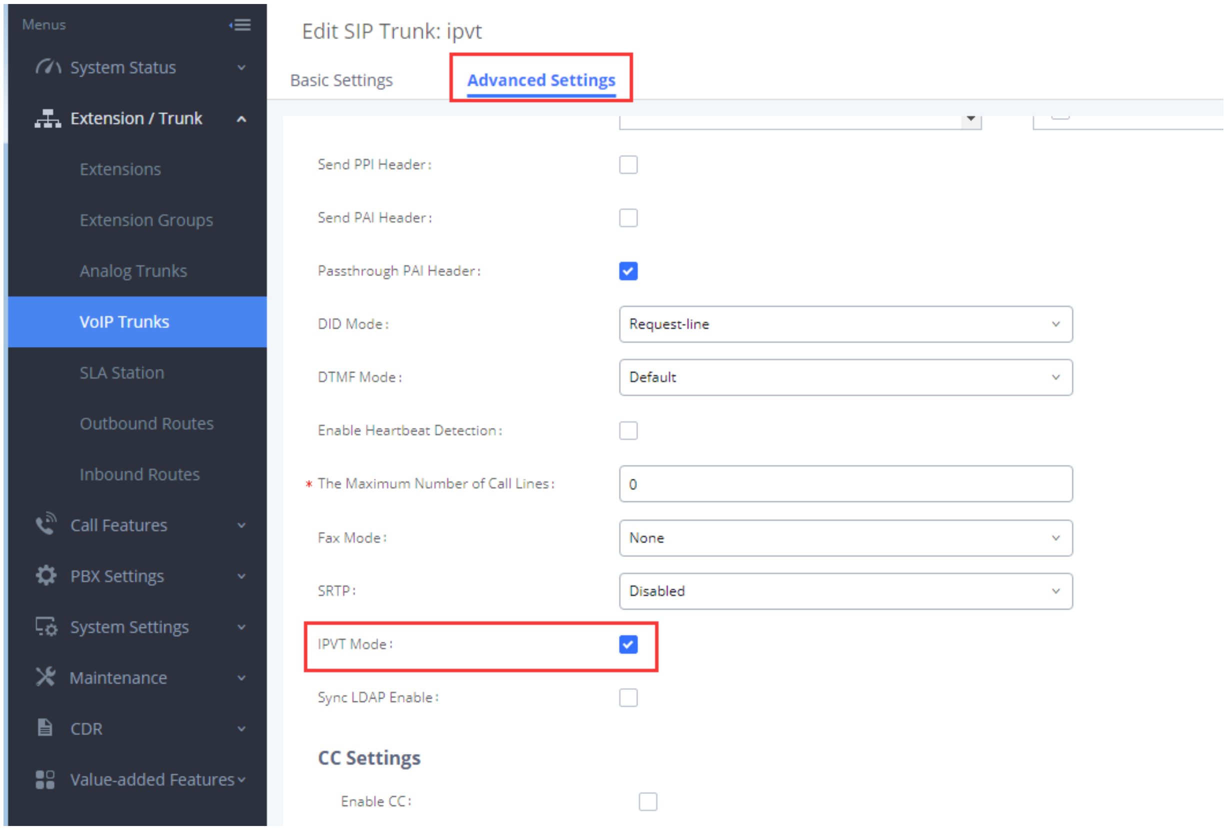

Figure 4: Configure IPVT Mode

Configure Outbound Route

Users could go to "Extension / Trunk → Outbound Routes", and click on "Add" to add the Outbound Route. As the figure shows below:

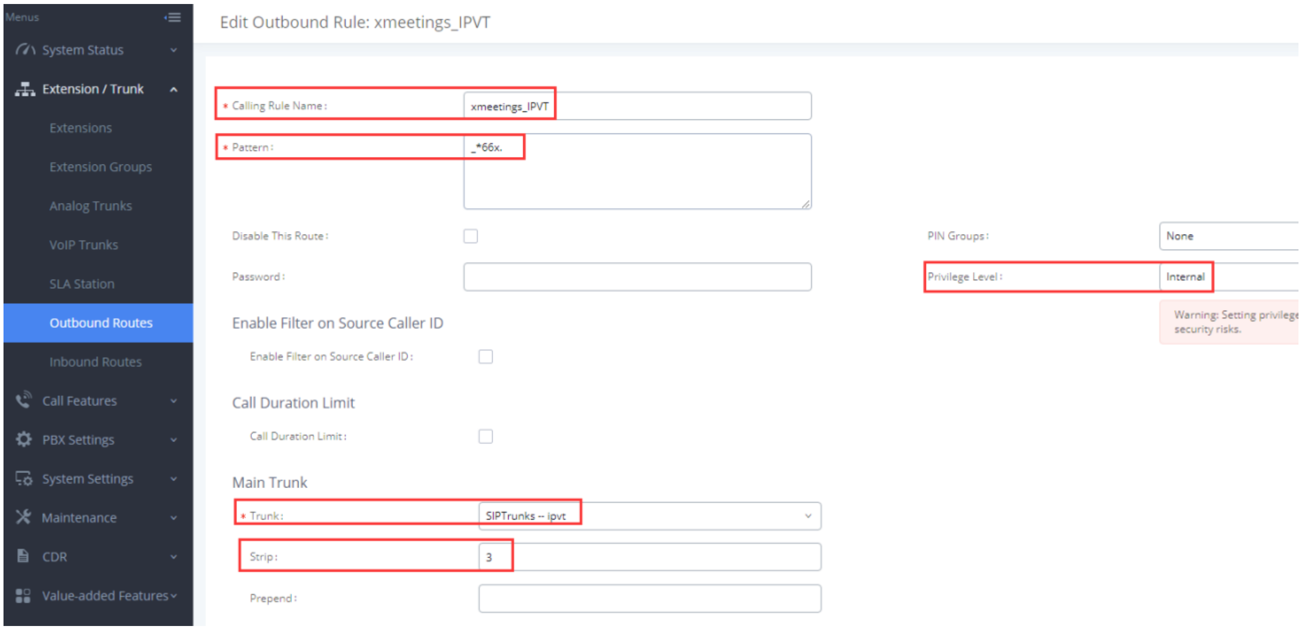

Figure 5: Configure Outbound Route

- **Configure Calling Rule Name:** Users need to fill in the Calling Rule Name for each Outbound Route, and the duplicated Calling Rule Name is not allowed.

- **Configure Pattern:** Users need to configure “Pattern” to recognize the dialing numbers for UCM, and the initial pattern should be “_”. The special characters and wildcard characters are allowed for patterns configuration. For instance, users could configure the pattern as “prefix + meeting ID”, such as “_*99x”. Then, UCM clients could dial “*99 + IPVideoTalk meeting ID” to dial into the meeting. The meeting ID could be 1 or multiple digits, and users may need to configure “Strip” option, please see the table below:

Table 1: Pattern Rule

Parameters	Description
X/x	0-9
Z/z	1-9
N/n	2-9

[345-9]	3,4,5,6,7,8,9
!	0 or multiple characters (any character)
.	1 or multiple characters (any character)

- **Configure Privilege Level:** Users need to configure the VoIP Trunk Privilege Level as "Internal" since the UCM clients' default privilege level is "Internal". The privilege level of the clients should be no lower than Outbound Route privilege level. Otherwise, the server will send 603 error messages to the clients.
- **Configure Use Trunk:** Users need to select the configured VoIP Trunk.
- **Configure Strip:** Users could configure the how many characters will be ignored for the prefix. For example, if users want to "*99", users could set "3" for this option.

Users could click on "Save" → "Apply" to create the new Outbound Route, as the figure shows below:

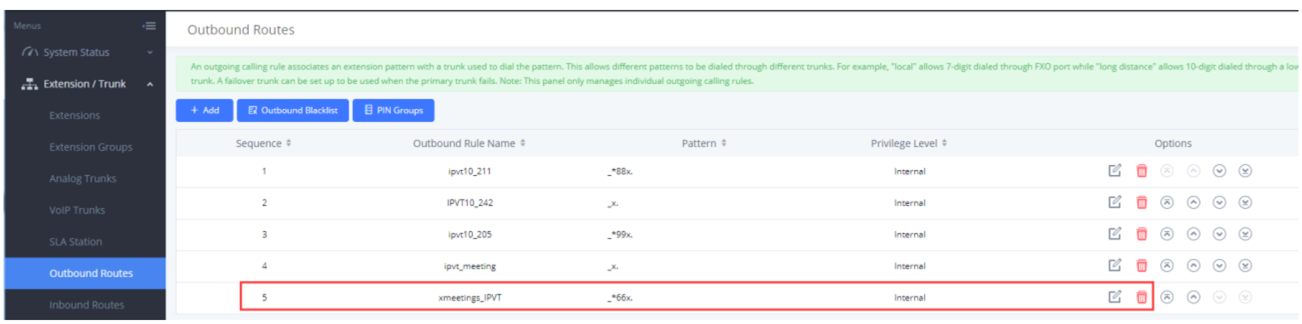

Figure 6: Create Outbound Route

Configure UCM Clients

Register SIP Account:

Log into the device's Web UI -> Account -> General Settings, and input the extension account information which is in UCM server.

SIP Sever: Fill in the UCM server address and SIP port number. Please note that the port should correspond to the SIP Transport type.

Account and Password: Input the extension account and password which are in UCM server.

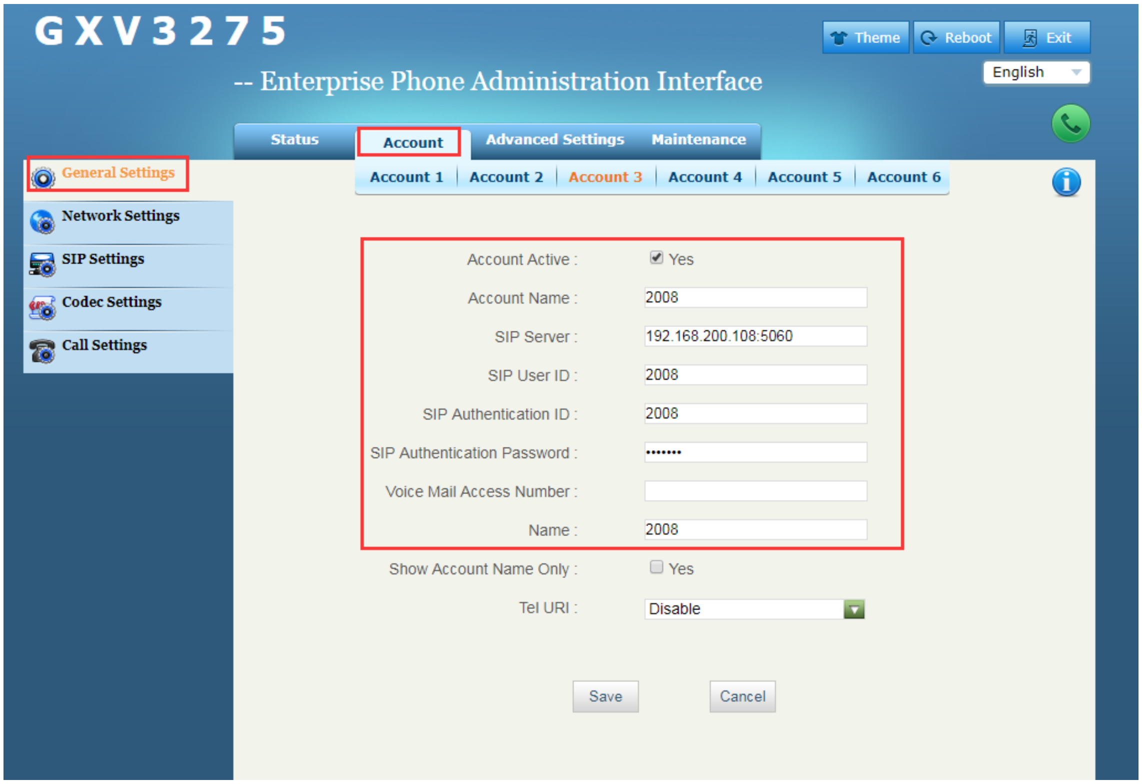

Figure 7: Configure General Settings on Clients

- **Configure Codecs:** Users could go to the UCM client's Web UI → Account → Codec Settings to select the codecs. Users have to select at least one same codec as the codec for the SIP account configured on the UCM clients.
- **Preferred Vocoder:** There should be at least one audio/video codec which is supported by both UCM client and IPVideoTalk server.

Note: We recommend enabling the option "Use First Matching Vocoder in 200OK SDP".

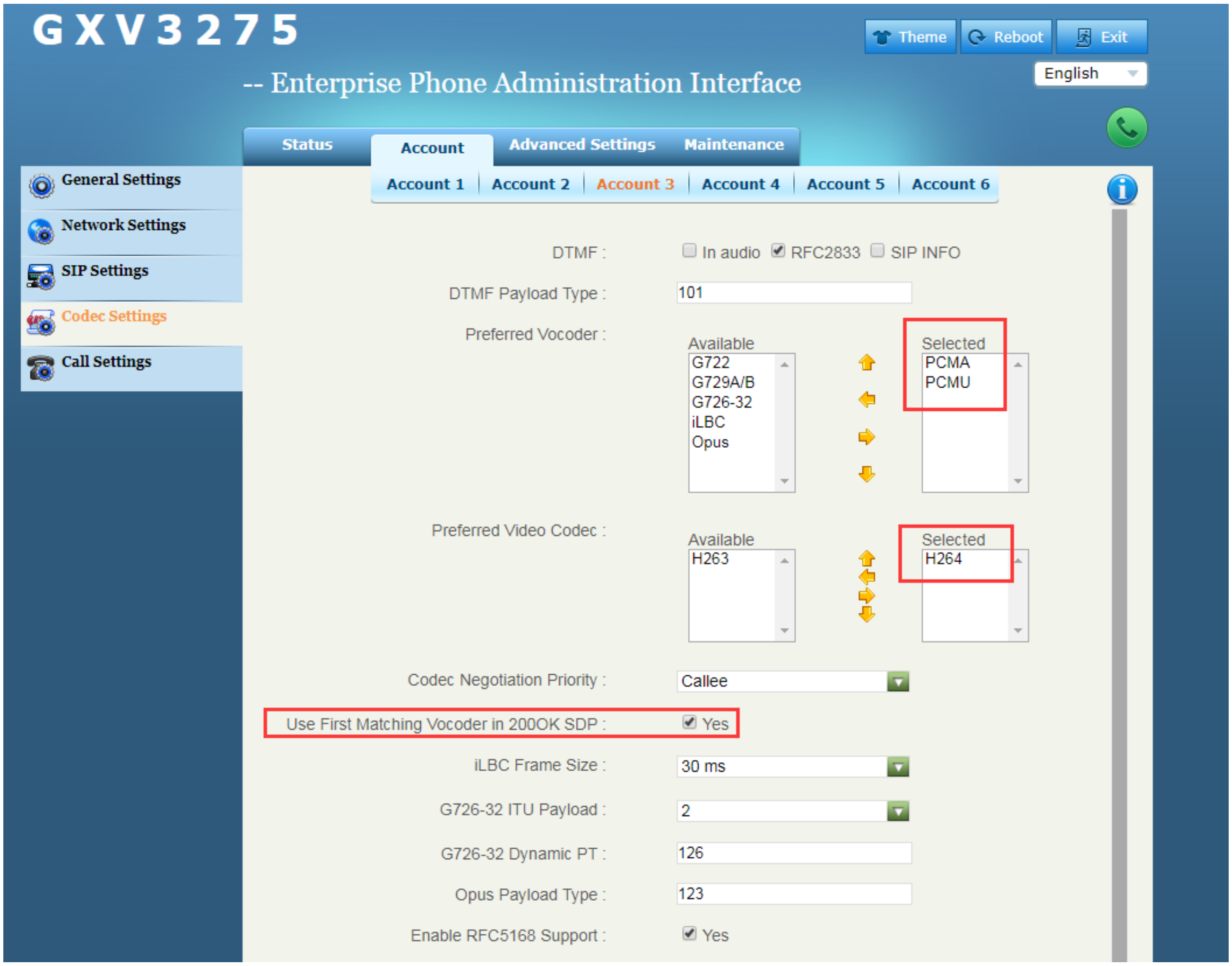

Figure 8: Configure Codecs on Clients

DIAL INTO IPVIDEOTALK MEETINGS

We assume client A has a registered IPVideoTalk ID, and client B has a registered SIP extension in UCM (e.g. 2008), the Dial Prefix for SIP Trunk is “*99”.

UCM Extension Joins into IPVideoTalk Meeting

Scenario :

UCM extension joins into the IPVideoTalk meeting by dialing IPVideoTalk meeting ID via audio call.

Prerequisite:

Active meeting M

Operations:

Users could dial IPVideoTalk meeting ID M (*99M) to join into the meeting on client B.

Conference Control

Prerequisite: Client A and client B are in the IPVideoTalk meeting M.

Operations:

1. Client A mutes/unmutes the audio.
2. Client B mutes/unmutes the audio.

Figure 9: Meeting Call using IPVideoTalk Service with the UCM