Grandstream Networks, Inc.

UCM6102/6104/6108/6116 IP PBX Appliance

User Manual

Grandstream Networks, Inc.

www.grandstream.com

UCM61xx User Manual

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CHANGE LOG

This section documents significant changes from previous versions of the UCM61xx user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

FIRMWARE VERSION 1.0.1.22

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream UCM6102/6104/6108/6116. UCM6102/6104/6108/6116 is an innovative IP PBX appliance designed for small to medium business. Powered by an advanced hardware platform with robust system resources, the UCM6102/6104/6108/6116 offers a highly versatile state-of-the-art Unified Communication (UC) solution for converged voice, video, data, fax and video surveillance application needs. Incorporating industry-leading features and performance, the UCM6102/6104/6108/6116 offers quick setup, deployment with ease and unrivaled reliability all at an unprecedented price point.

⚠ **Caution:**

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

⚠ **Warning:**

Please do not use a different power adaptor with the UCM6102/6104/6108/6116 as it may cause damage to the products and void the manufacturer warranty.

This document is subject to change without notice. The latest electronic version of this user manual is available for download here:
http://www.grandstream.com/support

PRODUCT OVERVIEW

FEATURE HIGHTLIGHTS

- 1GHz ARM Cortex A8 application processor, large memory (512MB DDR RAM, 4GB NAND Flash), and dedicated high performance multi-core DSP array for advanced voice processing.
- Integrated 2/4/8/16 PSTN trunk FXO ports, 2 analog telephone FXS ports with lifeline capability in case of power outage, and up to 50 SIP trunk options.
- Gigabit network port(s) with integrated PoE, USB, SD; integrated NAT router with advanced QoS support (UCM6102 only).
- Supports a wide range of popular voice codes (including G.711 A-law/U-law, G.722, G.723.1, G.726, G.729A/B, iLBC, GSM), video codec (including H.264, H.263, H.263+), and Fax (T.38).
- Hardware DSP based 128ms-tail-length carrier-grade line echo cancellation (LEC).
- Supports up to 500 SIP endpoint registration, up to 60 concurrent calls and up to 32 conference attendees.
- Flexible dial plan, call routing, site peering, call recording.
- Automated detection and provisioning of IP phones, video phones, ATA and other endpoints for easy deployment.
- Hardware encryption accelerator to ensure strongest security protection using SRTP, TLS, and HTTPS.

TECHNICAL SPECIFICATIONS

Table 1: Technical Specifications

Interfaces	
Analog Telephone FXS Ports	2 ports (both with lifetime capability in case of power outage)
PSTN Line FXO Ports	• UCM6102: 2 ports • UCM6104: 4 ports • UCM6108: 8 ports • UCM6116: 16 ports
Network Interfaces	• UCM6108/6116: Single 10M/100M/1000M RJ45 Ethernet port with integrated PoE Plug (IEEE 802.3at-2009) • UCM6102/6104: Dual 10M/100M/1000M RJ45 Ethernet ports with integrated PoE Plug (IEEE 802.3at-2009)
NAT Router	Yes, UCM6102 only
Peripheral Ports	USB, SD
LED Indicators	Power/Ready, Network, PSTN Line, USB, SD

LCD Display	128x32 graphic LCD with DOWN and OK button
Reset Switch	Yes
Voice/Video Capabilities	
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection and auto-switch to G.711
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC, GSM; T.38
Video Codecs	H.264, H.263, H.263+
QoS	Layer 3 QoS
Signaling and Control	
DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol and Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery and auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66/multicast SIP SUBSCRIBE/mDNS)
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Security	
Media	SRTP, TLS, HTTPS, SSH
Physical	
Universal Power Supply	• Output: 12VDC, 1.5A • Input: 100-240VAC, 50-60Hz
Environmental	• Operating: 32 - 104^{o}F / 0 - 40^{o}C, 10-90% (non-condensing) • Storage: 14 - 140^{o}F / -10 - 60^{o}C
Dimensions	• UCM6102/6104: 226mm (L) x 155mm (W) x 34.5mm (H) • UCM6108/6116: 440mm (L) x 185mm (W) x 44mm (H)
Mounting	• UCM6102/6104: Wall mount and Desktop • UCM6108/6116: Rack mount and Desktop
Additional Features	
Multi-language Support	Yes, English/Chinese/Spanish/French/German/Russian/Italian for Web GUI; Customizable IVR to support any language
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT, NTT Japan

Polarity Reversal/ Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability busy level, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response)
Concurrent Calls	• UCM6102: Up to 30 simultaneous calls • UCM6104: Up to 45 simultaneous calls • UCM6108/6116: Up to 60 simultaneous calls
Conference Bridges	• UCM6102/6104: Up to 3 password-protected conference bridges allowing up to 25 simultaneous PSTN or IP participants • UCM6108/6116: Up to 6 password-protected conference bridges allowing up to 32 simultaneous PSTN or IP participants
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom and etc
Compliance	• FCC: Part 15 (CFR 47) Class B, Part 68 • CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS • A-TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 adITU-T K.21 (Basic Level) • UL 60950 (power adapter)

INSTALLATION

Before deploying and configuring the UCM61xx, the device needs to be properly powered up and connected to network. This section describes detailed information on installation, connection and warranty policy of the UCM61xx.

EQUIPMENT PACKAGING

Table 2: UCM6102/UCM6104 Equipment Packaging

Main Case	Yes (1)
Power Adaptor	Yes (1)
Ethernet Cable	Yes (1)
Quick Installation Guide	Yes (1)

Table 3: UCM6108/UCM6116 Equipment Packaging

Main Case	Yes (1)
Power Adaptor	Yes (1)
Ethernet Cable	Yes (1)
Quick Installation Guide	Yes (1)
Wall Mount	Yes (2)
Screws	Yes (6)

CONNECT YOUR UCM61XX

CONNECT THE UCM6102

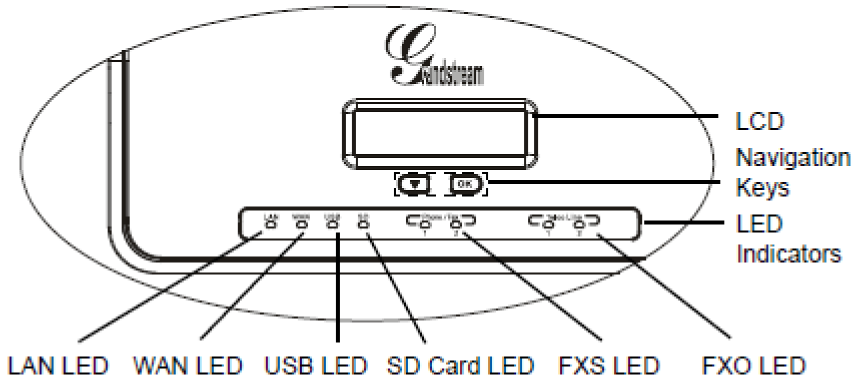


Figure 1: UCM6102 Front View

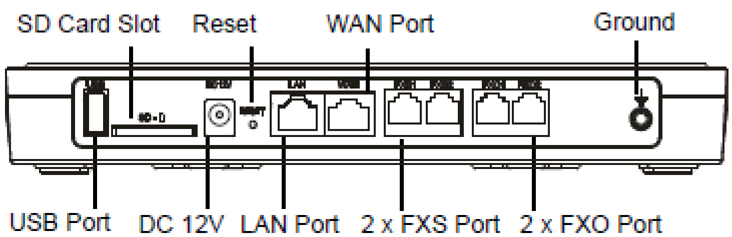


Figure 2: UCM6102 Back View

To set up the UCM6102, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the UCM6102;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6102. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Wait for the UCM6102 to boot up. The LCD in the front will show the device hardware information when the boot process is done;
5. Once the UCM6102 is successfully connected to network, the LED indicator for WAN in the front will be in solid green and the LCD shows up the IP address;
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

CONNECT THE UCM6104

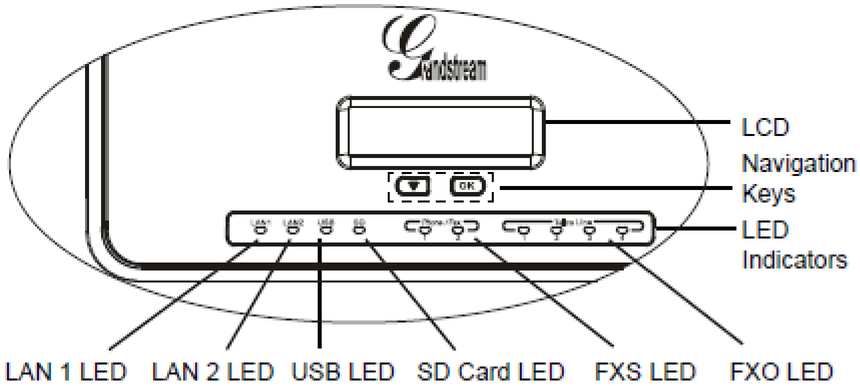


Figure 3: UCM6104 Front View

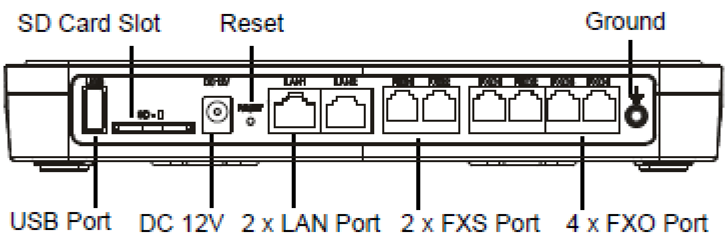


Figure 4: UCM6104 Back View

To set up the UCM6104, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the LAN 1 port of the UCM6104;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6104. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Wait for the UCM6104 to boot up. The LCD in the front will show the device hardware information when the boot process is done;
5. Once the UCM6104 is successfully connected to network, the LED indicator for LAN 1 in the front will be in solid green and the LCD shows up the IP address;
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

CONNECT THE UCM6108

To set up the UCM6108, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the LAN port of the UCM6108;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6108. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Wait for the UCM6108 to boot up. The LCD in the front will show the device hardware information when the boot process is done;
5. Once the UCM6108 is successfully connected to network, the LED indicator for NETWORK in the front will be in solid green and the LCD shows up the IP address;
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

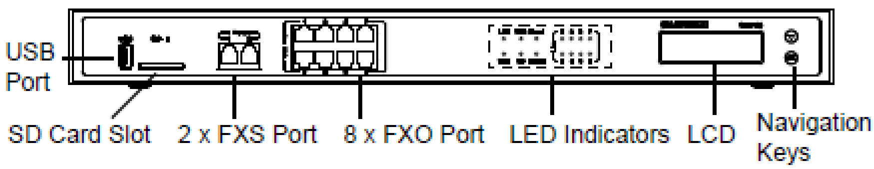


Figure 5: UCM6108 Front View

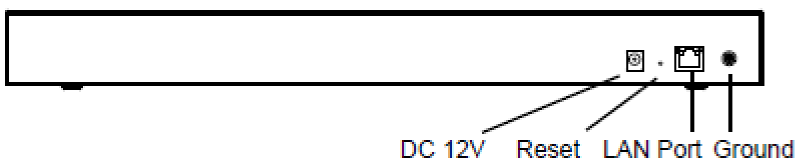


Figure 6: UCM6108 Back View

CONNECT THE UCM6116

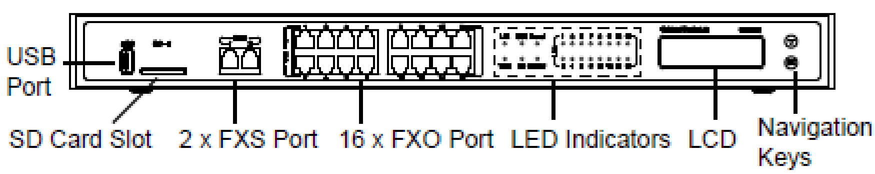


Figure 7: UCM6116 Front View

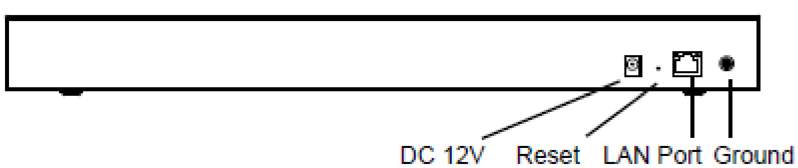


Figure 8: UCM6116 Back View

To set up the UCM6116, follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the LAN port of the UCM6116;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the UCM6116. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Wait for the UCM6116 to boot up. The LCD in the front will show the device hardware information when the boot process is done;
5. Once the UCM6116 is successfully connected to network, the LED indicator for NETWORK in the front will be in solid green and the LCD shows up the IP address;
6. (Optional) Connect PSTN lines from the wall jack to the FXO ports; connect analog lines (phone and Fax) to the FXS ports.

SAFETY COMPLIANCES

The UCM61xx complies with FCC/CE and various safety standards. The UCM61xx power adapter is compliant with the UL standard. Use the universal power adapter provided with the UCM61xx package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

WARRANTY

If the UCM61xx was purchased from a reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Grandstream, contact our Technical Support Team for a RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

⚠ **Warning:**

Use the power adapter provided with the UCM61xx. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

GETTING STARTED

The UCM61xx provides LCD interface, LED indication and web GUI configuration interface.

- The LCD displays hardware, software and network information of the UCM61xx. Users could also navigate in the LCD menu for device information and basic network configuration.
- The LED indication at the front of the device provides interface connection and activity status.
- The web GUI gives users access to all the configurations and options for UCM61xx setup.

This section provides step-by-step instructions on how to use the LCD menu, LED indicators and Web GUI of the UCM61xx. Once the basic settings are done, users could start making calls from UCM61xx extension registered on a SIP phone as described at the end of this section.

USE THE LCD MENU

- **Default LCD Display**
  By default, when the device is powered up, the LCD will show device model (e.g., UCM6116), hardware version (e.g., V1.5A) and IP address. Press "Down" button and the system time will be displayed as well.

- **Menu Access**
  Press "OK" button to start browsing menu options. Please see menu options in [***Table 4: LCD Menu Options***].

- **Menu Navigation**
  Press the "Down" arrow key to browser different menu options. Press the "OK" button to select an entry.

- **Exit**
  If "Back" option is available in the menu, select it to go back to the previous menu. For "Device Info" "Network Info" and "Web Info" which do not have "Back" option, simply press the "OK" button to go back to the previous menu. Also, the LCD will display default idle screen after staying in menu option for 15 seconds.

- **LCD Backlight**
  The LCD backlight will be on upon key pressing. The backlight will go off after the LCD stays in idle for 30 seconds.

The following table shows the LCD menu options.

Table 4: LCD Menu Options

View Events	• **Critical Events** • **Other Events**
Device Info	• **Hardware**: Hardware version number • **Software**: Software version number • **P/N**: Part number • **WAN MAC**: WAN side MAC address (UCM6102 only) • **LAN MAC**: LAN side MAC address • **Uptime**: System up time
Network Info	For UCM6104/UCM6108/UCM6116: • **LAN Mode**: DHCP, Static IP, or PPPoE • **LAN IP**: IP address • **LAN Subnet Mask** For UCM6102: • **WAN Mode**: DHCP, Static IP, or PPPoE • **WAN IP**: IP address • **WAN Subnet Mask** • **LAN IP**: IP address • **LAN Subnet Mask**
Network Menu	For UCM6104/UCM6108/UCM6116: • **LAN Mode**: Select LAN mode as DHCP, Static IP or PPPoE For UCM6102: • **WAN Mode**: Select WAN mode as DHCP, Static IP or PPPoE
Factory Menu	• **Reboot** • **Factory Reset** • **LCD Test Patterns** Press "Down" button to test different LCD patterns. When done, press "OK" button to exit. • **Fan Mode** Select "Auto" or "On". • **LED Test Patterns**

	Select "All On" "All Off" or "Blinking" and check LED status. • **RTC Test Patterns** Select "2022-02-22 22:22" or "2011-01-11 11:11" to start the RTC (Real-Time Clock) test pattern. Then check the system time from LCD idle screen by pressing "DOWN" button, or from web GUI->**System Status**->**General** page. Reboot the device manually after the RTC test is done. • **Hardware Testing** Select "Test SVIP" to perform SVIP test on the device. This is mainly for factory testing purpose which verifies the hardware connection inside the device. The diagnostic result will display in the LCD after the test is done.
Web Info	• **Protocol**: Web access protocol. HTTP or HTTPS. By default it's HTTPS • **Port**: Web access port number. By default it's 8089

USE THE LED INDICATORS

The UCM61xx has LED indicators in the front to display connection status. The following table shows the status definitions.

Table 5: UCM6102/UCM6104 LED INDICATORS

LED Indicator	LED Status
LAN WAN USB SD FXS (Phone/Fax) FXO (Telco Line)	**Solid**: Connected **Flashing**: Data Transferring **OFF**: Not Connected

Table 6: UCM6108/UCM6116 LED INDICATORS

LED	LED Status
NETWORK	**Solid**: Connected **OFF**: Not Connected
ACT USB SD Phone (FXS) Line (FXO)	**Solid:** Connected **Flashing**: Data Transferring **OFF**: Not Connected

USE THE WEB GUI

ACCESS WEB GUI

The UCM61xx embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the device through a Web browser such as Microsoft's IE, Mozilla Firefox, Google Chrome and etc.

Figure 9: UCM6116 Web GUI Login Page

To access the Web GUI:

1. Connect the computer to the same network as the UCM61xx;
2. Ensure the device is properly powered up and shows its IP address on the LCD;
3. Open a Web browser on the computer and enter the web GUI URL in the following format:

   http(s)://IP-Address:Port

   where the ***IP-Address*** is the IP address displayed on the UCM61xx LCD.

   By default, the protocol is HTTPS and the Port number is 8089.

   For example, if the LCD shows 192.168.40.167, please enter the following in your web browser:

   https://192.168.40.167:8089

4. Enter the administrator’s login and password to access the Web Configuration Menu. The default administrator's username and password is "admin" and "admin". It is highly recommended to change the default password after login for the first time.

WEB GUI CONFIGURATIONS

There are four main sections in the Web GUI for users to view the PBX status, configure and manage the PBX.

- **Status:** Displays PBX status, System Status and CDR.
- **PBX:** To configure extensions, trunks, call routes, zero config for auto provisioning, call features, internal options, IAX settings and SIP settings.
- **Settings:** To configure network settings, firewall settings, change password, LDAP Server, HTTP Server, Email Settings and Time Settings.
- **Maintenance**: To perform firmware upgrade, backup configurations, cleaner setup, reset/reboot, syslog setup and troubleshooting.

WEB GUI LANGUAGES

Currently the UCM61xx web GUI supports the following languages:

English
Chinese
Spanish
French
Portuguese
Russian
Italian
Polish
German

Users can select the displayed language in web GUI login page, or at the upper right of the web GUI after logging in.

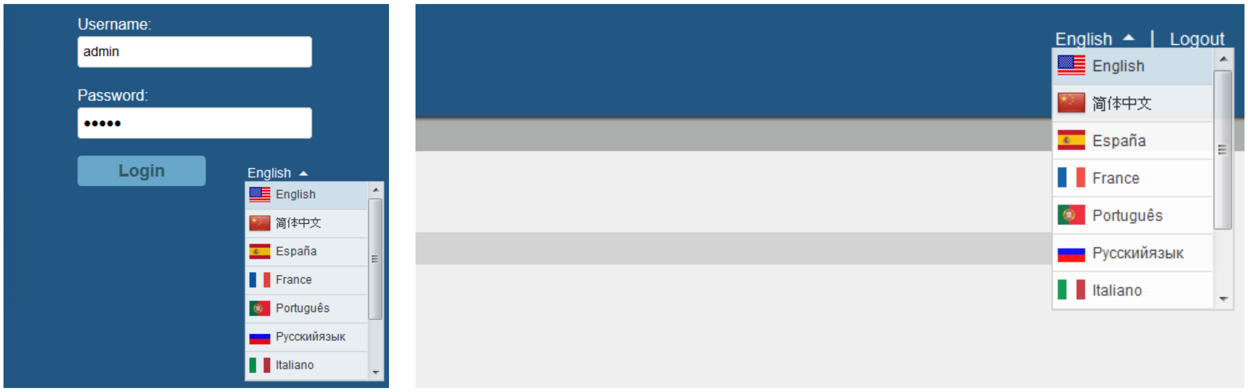


Figure 10: UCM61xx Web GUI Language

SAVE AND APPLY CHANGES

Click on "Save" button after configuring the web GUI options in one page. After saving all the changes, make sure click on "Apply Changes" button on the upper right of the web page to submit all the changes. If the change requires reboot to take effect, a prompted message will pop up for you to reboot the device.

MAKE YOUR FIRST CALL

Power up the UCM61xx and your SIP end point phone and connect both to network. Then follow the steps below to make your first call.

1. Log in the UCM61xx web GUI, go to **PBX**->**Basic/Call Routes**->**Extensions**;
2. Click on "Create New User" to create a new extension. You will need User ID, Password and Voicemail Password information to register and use the extension later;
3. Register the extension on your phone with the SIP User ID, SIP server and SIP Password information. The SIP server address is the UCM61xx IP address;
4. When your phone is registered with the extension, dial *97 to access the voicemail box. Enter the Voicemail Password once you hear "Password" voice prompt;
5. Once successfully logged in, you will be prompted with the Voice Mail Main menu;
6. You are successfully connected to the PBX system now.

SYSTEM SETTINGS

This section explains configurations for system-wide parameters on the UCM61xx. Those parameters include Network Settings, Firewall, Change Password, LDAP server, HTTP server, Email settings and Time Settings.

NETWORK SETTINGS

After successfully connecting the UCM61xx to the network for the first time, users could login the Web GUI and go to **Settings**->**Network Settings** to configure the network parameters for the device.

The network setting options are similar for UCM6108 and UCM6116. Additional network functions and settings are available for UCM6102 and UCM6104:

- UCM6102 supports Router/Switch/Dual mode functions;
- UCM6104 supports Switch/Dual mode functions.

In this section, all the available network setting options are listed for each model. Select each tab in web GUI->**Settings**->**Network Settings** page to configure LAN settings, WAN settings (UCM6102 only), 802.1X and Port Forwarding (UCM6102 only).

BASIC SETTINGS

Please refer to the following tables for basic network configuration parameters on UCM6102, UCM6104, and UCM6108/UCM6116 respectively.

Table 7: UCM6102 Network Settings->Basic Settings

Method	Select "Route", "Switch" or "Dual" mode on the network interface of UCM6102. The default setting is "Route". • **Route** WAN port interface will be used for uplink connection. LAN port interface will be used to serve as router. • **Switch** WAN port interface will be used for uplink connection. LAN port interface will be used as bridge for PC connection. • **Dual**

	Both ports can be used for uplink connection. Users will need assign the default interface in option "Default Interface".
Preferred DNS Server	Enter the preferred DNS server address.
WAN (when "Method" is set to "Route")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
LAN (when Method is set to "Route")	
IP Address	Enter the IP address assigned to LAN port. The default setting is 192.168.2.1.
Subnet Mask	Enter the subnet mask. The default setting is 255.255.255.0.
DHCP Server Enable	Enable or disable DHCP server capability. The default setting is "Yes".
DNS Server 1	Enter DNS server address 1. The default setting is 8.8.8.8.
DNS Server 2	Enter DNS server address 2. The default setting is 208.67.222.222.
Allow IP Address From	Enter the DHCP IP Pool starting address. The default setting is 192.168.2.100.
Allow IP Address To	Enter the DHCP IP Pool ending address. The default setting is 192.168.2.254.
Default IP Lease Time	Enter the IP lease time (in seconds). The default setting is 43200.
LAN (when Method is set to "Switch")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.
LAN 1 / LAN 2 (when Method is set to "Dual")	

Default Interface	If "Dual" is selected as "Method", users will need assign the default interface to be LAN 1 (mapped to UCM6102 WAN port) or LAN 2 (mapped to UCM6102 LAN port) and then configure network settings for LAN1/LAN2. The default interface is LAN 1.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.

Table 8: UCM6104 Network Settings->Basic Settings

Method	Select "Switch" or "Dual" mode on the network interface of UCM6104. The default setting is "Switch". • **Switch** LAN 1 port interface will be used for uplink connection. LAN 2 port interface will be used as bridge for PC connection. • **Dual** Both ports can be used for uplink connection. Users will need assign the default interface in option "Default Interface".
Preferred DNS Server	Enter the preferred DNS server address.
LAN (when Method is set to "Switch")	
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.

LAN 1 / LAN 2 (when Method is set to "Dual")	
Default Interface	If "Dual" is selected as "Method", users will need assign the default interface to be LAN 1 or LAN 2. The default interface is LAN 1.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings. The default setting is 192.168.0.160.
Gateway IP	Enter the gateway IP address for static IP settings. The default setting is 0.0.0.0.
Subnet Mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.0.0.
DNS Server 1	Enter the DNS server 1 address for static IP settings. The default setting is 0.0.0.0.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.

Table 9: UCM6108/UCM6116 Network Settings->Basic Settings

Preferred DNS Server	Enter the preferred DNS server address.
IP Method	Select DHCP, Static IP, or PPPoE. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings.
Gateway IP	Enter the gateway IP address for static IP settings.
Subnet Mask	Enter the subnet mask address for static IP settings.
DNS Server 1	Enter the DNS server 1 address for static IP settings.
DNS Server 2	Enter the DNS server 2 address for static IP settings.
User Name	Enter the user name to connect via PPPoE.
Password	Enter the password to connect via PPPoE.

802.1X

The UCM61xx provides users 802.1X settings for LAN port and WAN port (UCM6102 only).

Table 10: UCM61xx Network Settings->802.1X

802.1X Mode	Select 802.1X mode. The default setting is "Disable". The supported 802.1X mode are: • EAP-MD5 • EAP-TLS • EAP-PEAPv0/MSCHAPv2

Identity	Enter 802.1X mode identity information.
MD5 Password	Enter 802.1X mode MD5 password information.
802.1X Certificate	Select 802.1X certificate from local PC and then upload.
802.1X Client Certificate	Select 802.1X client certificate from local PC and then upload.

PORT FORWORDING (UCM6102 ONLY)

The UCM6102 network interface supports router functions which provides users the ability to do port forwarding. Please see port forwarding settings in the table below.

Table 11: UCM6102 Network Settings->Port Forwarding

WAN Port	Specify the WAN port number. Up to 8 ports can be configured.
LAN IP	Specify the LAN IP address.
LAN Port	Specify the LAN port number.
Protocol Type	Select protocol type "UDP Only", "TCP Only" or "TCP/UDP" for the forwarding in the selected port. The default setting is "UDP Only".

FIREWALL

The UCM61xx provides users firewall configurations to prevent certain malicious attack to the UCM61xx system. Users could configure to allow, restrict or reject specific traffic through the device for security and bandwidth purpose. To configure firewall settings in UCM61xx, go to Web GUI->**Settings**->**Firewall** page.

STATIC DEFENSE

Under Web GUI->**Settings**->**Firewall->Static Defense** page, users will see the following information:

- Current service information with port, process and type
- Typical firewall settings
- Custom firewall settings

The following table shows a sample current service status running on UCM61xx.

Table 12: UCM61xx Firewall->Static Defense->Current Service

Port	Process	Type	Protocol or Service
7777	asterisk	tcp/IPv4	SIP
389	slapd	tcp/IPv4	LDAP
22	dropbear	tcp/IPv4	SSH
80	lighthttpd	tcp/IPv4	HTTP
8089	lighthttpd	tcp/IPv4	HTTPS
69	opentftpd	udp/IPv4	TFTP
9090	asterisk	udp/IPv4	SIP
6060	zero_config	udp/IPv4	UCM61xx zero_config service
5060	asterisk	udp/IPv4	SIP
4569	asterisk	udp/IPv4	SIP
5353	zero_config	udp/IPv4	UCM61xx zero_config service
37435	syslogd	udp/IPv4	Syslog

For typical firewall settings, users could configure the following options on the UCM61xx.

Table 13: Typical Firewall Settings

Ping Defense Enable	If enabled, ICMP response will not be allowed for Ping request. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6102 only) interface.
SYN-Flood Defense Enable	Enable to prevent SYN Flood denial-of-service attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6102 only) interface.
Death-of-Ping Defense Enable	Enable to prevent Death-of-Ping attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (UCM6102 only) interface.

Under "Custom Firewall Settings", users could create new rules to accept, reject or drop certain traffic going through the UCM61xx. To create new rule, click on "Create New Rule" button and a new window will pop up for users to specify rule options.

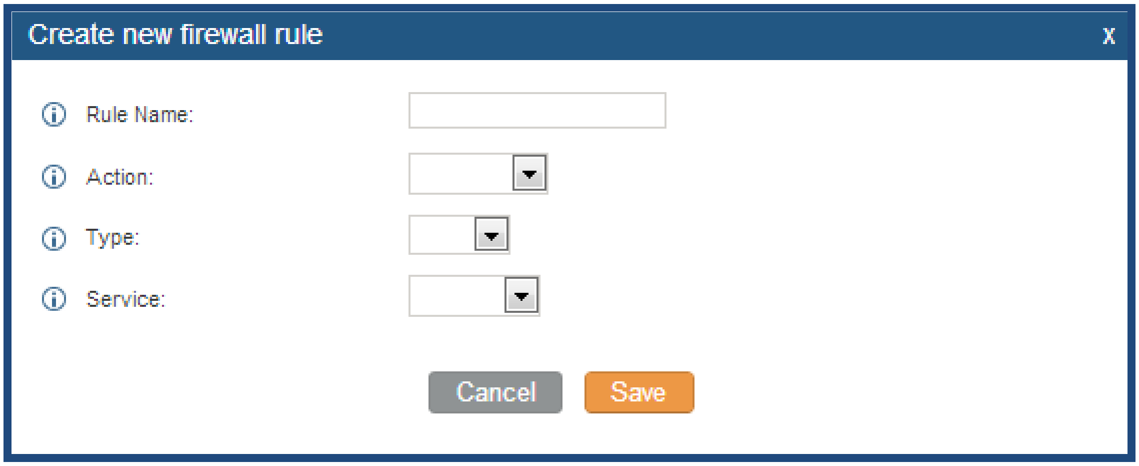


Figure 11: Create New Firewall Rule

Table 14: Firewall Rule Settings

Rule Name	Specify the Firewall rule name to identify the firewall rule.
Action	Select the action for the Firewall to perform. • ACCEPT • REJECT • DROP
Type	Select the traffic type. • IN If selected, users will need specify the network interface "LAN" or "WAN" (for UCM6102 only) for the incoming traffic. • OUT
Service	Select the service type. • FTP • SSH • Telnet • TFTP • HTTP • LDAP • Custom If selected, users will need specify Source (IP and port), Destination (IP and port) and Protocol (TCP, UDP or Both) for the service.

Save the change and click on "Apply" button. Then submit the configuration by clicking on "Apply Changes" on the upper right of the web page. The new rule will be listed at the bottom of the page with sequence number, rule name, action, protocol, type, source, destination and operation. Users can click on to edit the rule, or select to delete the rule.

DYNAMIC DEFENSE

The UCM61xx supports firewall dynamic defense that can blacklist hosts dynamically. It monitors the traffic coming into the UCM61xx and helps prevent massive connection attempts or brute force attacks to the device. The blacklist can be created and updated by the UCM61xx firewall, which will then be displayed in the web page. Please refer to the following table for dynamic defense options on UCM61xx.

Table 15: Firewall Dynamic Defense

Dynamic Defense Enable	Enable dynamic defense on UCM61xx firewall.
Periodical Time	Configure the dynamic defense periodic time interval (in minutes). If the number of TCP connections from a host exceeds the connection threshold within this period, this host will be added into Blacklist. The valid value is between 1 to 59 when dynamic defense is turned on.
Blacklist Update Interval	Configure the blacklist update time interval (in seconds). The default setting is 120 seconds.
Connection Threshold	Configure the connection threshold. Once the number of connections from the same host reaches the threshold, it will be added into the blacklist.
Dynamic Defense Whitelist	Configure the dynamic defense whitelist.

CHANGE PASSWORD

After login the Web GUI for the first time, it is highly recommended for users to change the default password "admin" to a more complicated password for security purpose. Follow the steps below to change the Web GUI access password.

- Go to Web GUI->**Settings**->**Change Password** page;
- Enter the old password first;
- Enter the new password and retype the new password to confirm. The new password field has to be at least 5 characters;
- Click on "Save" and the user will be logged out;

- Once the web page comes back to the login page again, enter the username "admin" and the new password to login.

LDAP SERVER

The UCM61xx has an embedded LDAP server for users to manage corporate phonebook in a centralized manner. By default, the LDAP server has generated the phonebook based on the created extensions already. If users have the Grandstream phone provisioned by the UCM61xx, the LDAP directory has been set up on the phone and can be used right away. Also, users could manually configure the LDAP client settings accordingly to manipulate the built-in LDAP server on the PBX.

To access LDAP Server settings, go to **Web GUI**->**Settings**->**LDAP Server**.

LDAP SERVER CONFIGURATIONS

The following figure shows the default LDAP server configurations on the UCM61xx.

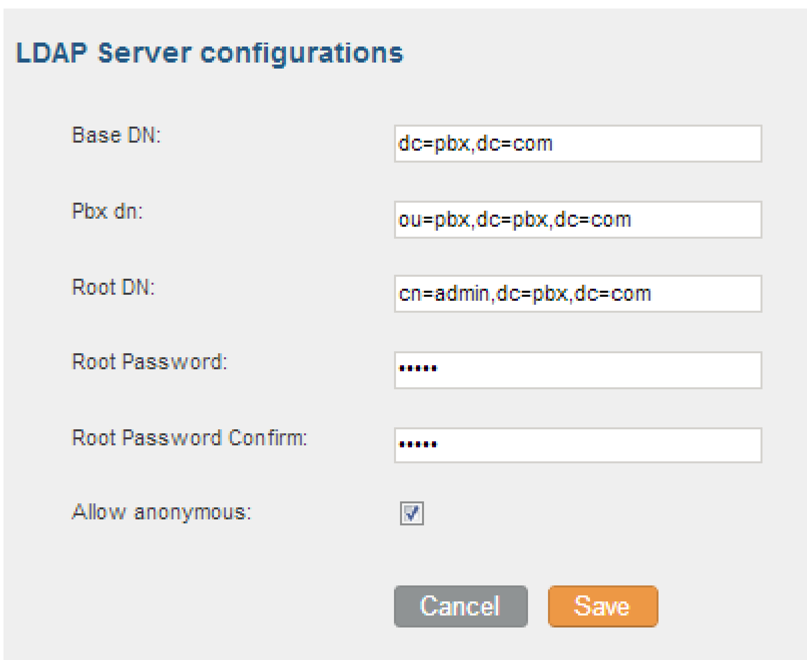
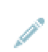


Figure 12: LDAP Server Configurations

The default phonebook list in this LDAP server can be viewed and edited by clicking on  of this Phonebook (the first phonebook under LDAP Phonebook).

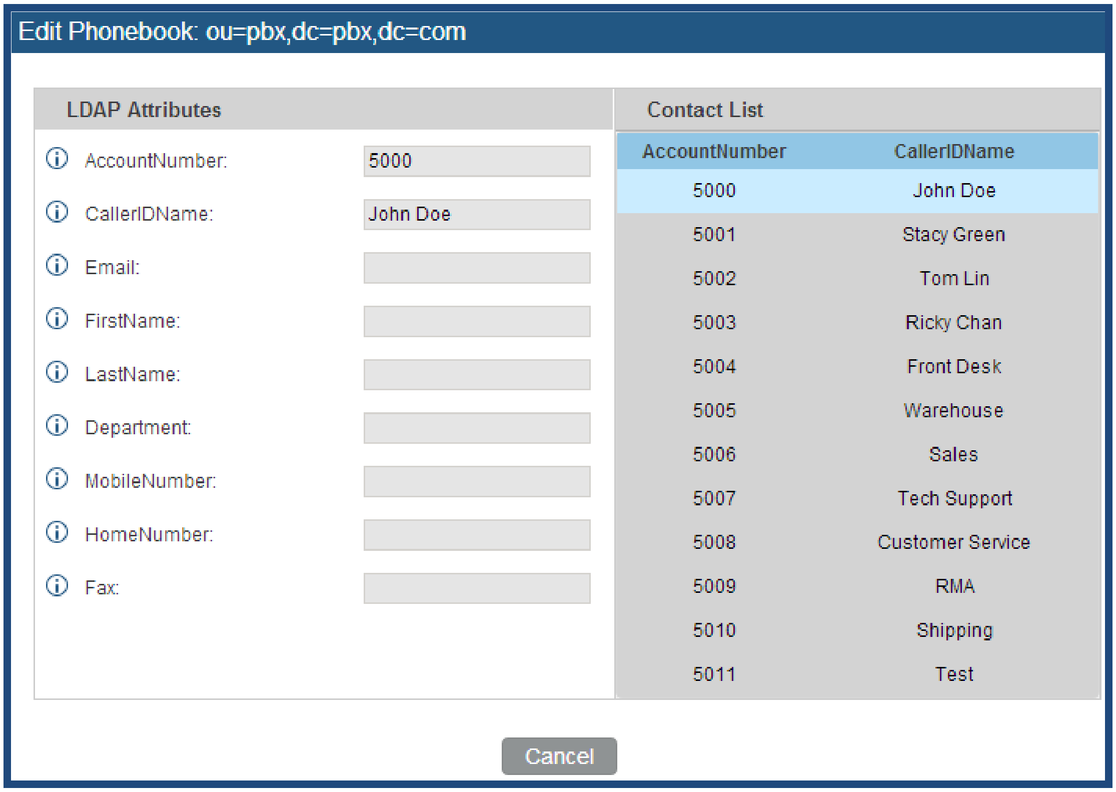


Figure 13: Default LDAP Phonebook in UCM61xx

LDAP PHONEBOOK

Users could use the default phonebook, edit the default phonebook as well as add new phonebook on the LDAP server. The first phonebook with default phonebook dn "ou=pbx,dc=pbx,dc=com" displayed on the LDAP server page is for extensions in this PBX. Users cannot add or delete contacts directly. The contacts information will need to be modified via Web GUI->**PBX**->**Basic/Call Routes**->**Extensions** first. The default LDAP phonebook will then be updated automatically.

A new sibling phonebook of the default PBX phonebook can be added by clicking on "Add" under "LDAP Phonebook" section.

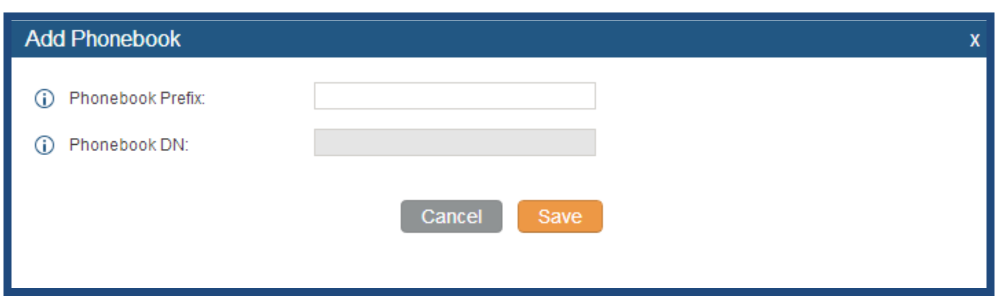


Figure 14: Add LDAP Phonebook

Once added, users can select to edit the phonebook attributes and contact list (see figure below), or select to delete the phonebook.

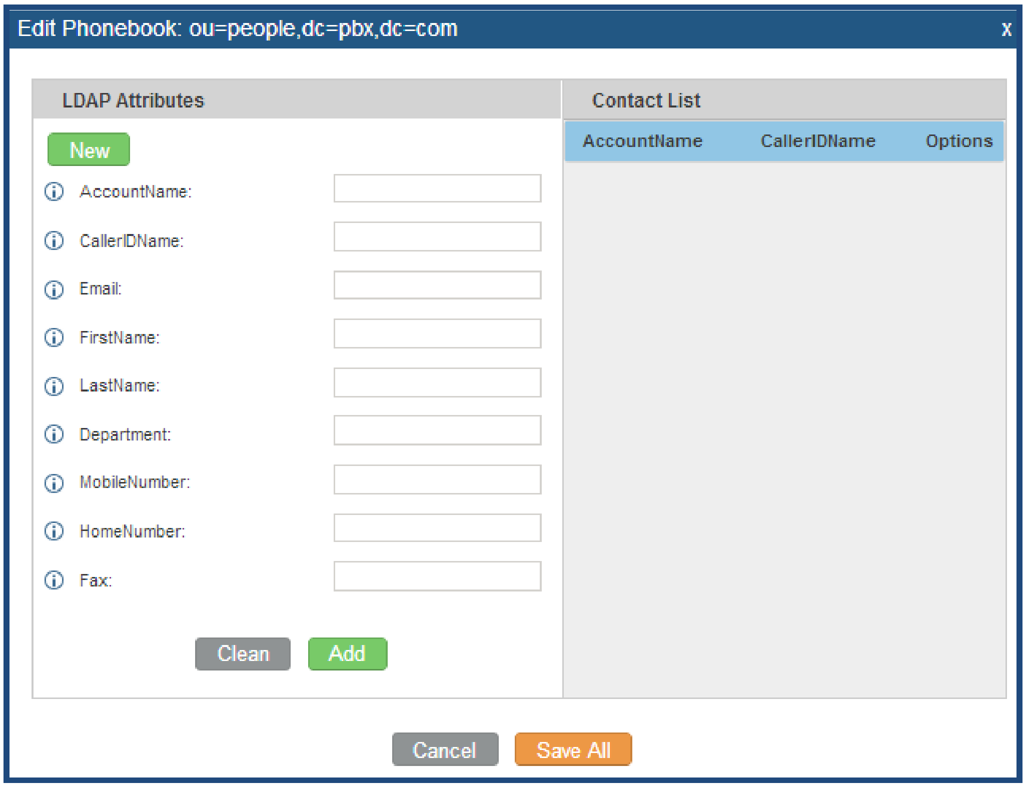


Figure 15: Edit LDAP Phonebook

LDAP CLIENT CONFIGURATIONS

The configuration on LDAP client is similar when you use other LDAP servers. Here we provide an example on how to configure the LDAP client on the SIP end points to use the default PBX phonebook. Please follow the instructions in the "LDAP Client Configurations" section (described below).

Suppose your server Base DN is "dc=Grandstream", your extension number is 1000 and your LDAP entry password is "1000", configure your LDAP client as follows (case insensitive):

Base DN: dc=Grandstream
Root DN: AccountName=1000,dc=Grandstream
Password: 1000
Filter: (&(CallerIDName=*)(AccountName=*))
Port: 389

The following figure shows the configuration information on a Grandstream GXP2200 to successfully use the LDAP server as configured in ***Figure 12: LDAP Server Configurations***.

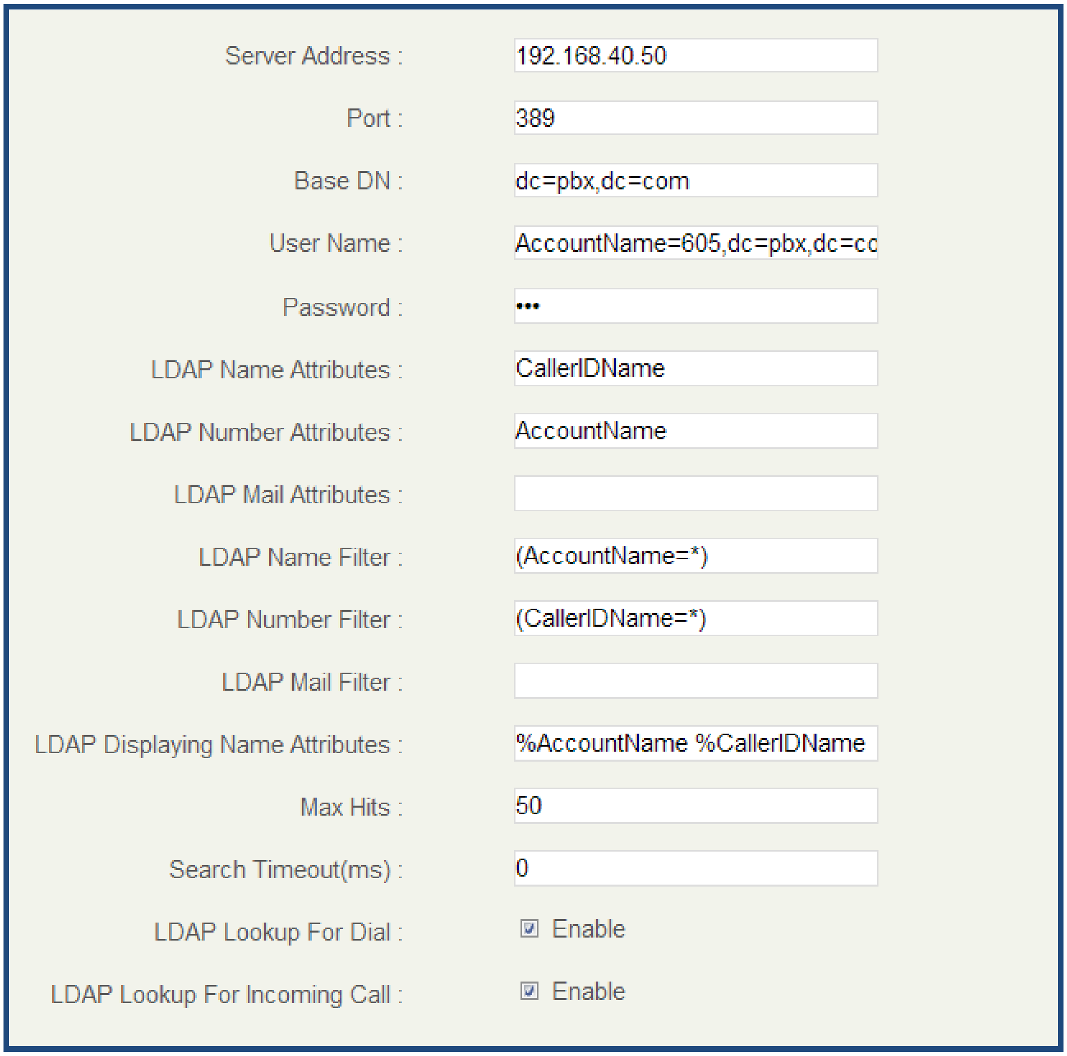


Figure 16: GXP2200 LDAP Phonebook Configuration

HTTP SERVER

The UCM61xx embedded web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow the users to configure the PBX through a Web browser such as Microsoft's IE, Mozilla Firefox and Google Chrome. By default, the PBX can be accessed via HTTPS using Port 8089 (e.g., https://192.168.40.50:8089). Users could also change the access protocol and port as preferred under Web GUI->**Settings**->**HTTP Server**.

Table 16: HTTP Server Settings

Redirect From Port 80	Enable or disable redirect from port 80. On the PBX, the default access protocol is HTTPS and the default port number is 8089. When this option is enabled, the access using HTTP with Port 80 will be redirected to HTTPS with Port 8089. The default setting is "Enable".
Protocol Type	Select HTTP or HTTPS. The default setting is "HTTPS".
Port	Specify port number to access the HTTP server. The default port number is 8089.

Once the change is saved, the web page will be redirected to the login page using the new URL. Enter the username and password to login again.

EMAIL SETTINGS

The Email application on the UCM61xx can be used to send out Emails to users with Fax (e.g., Fax-To-Email), Voicemail (Voicemail-To-Email) and other information as attachment. The configuration parameters can be accessed via Web GUI->**Settings**->**Email Settings**.

Table 17: Email Settings

TLS Enable	Enable or disable TLS during transferring/submitting your Email to other SMTP server. The default setting is "Yes".
Type	Select Email type. • MTA: Mail Transfer Agent. The Email will be sent from the configured domain. When MTA is selected, there is no need to set up SMTP server for it or no user login is required. However, the Emails sent from MTA might be considered as spam by the target SMTP server. • Client: Submit Emails to the SMTP server. A SMTP server is required and users need login with correct credentials.
Domain	Specify the domain name to be used in the Email.
Display Name	Specify the display name in the FROM header in the Email.
Sender	Specify the sender's Email address. For example, pbx@example.mycompany.com.

TIME SETTINGS

The current system time on UCM61xx is displayed under Web GUI->**Status**->**System Status**. To change the time settings on the UCM61xx, go to Web GUI->**Settings**->**Time Settings**.

Table 18: Time Settings

NTP Server	Specify the URL or IP address of the NTP server for the UCM61xx to synchronize the date and time. The default NTP server is ntp.ipvideotalk.com.
Enable DHCP Option 2	If set to "Yes", the UCM61xx is allowed to get provisioned for Time Zone from DHCP Option 2 in the local server automatically. The default setting is "Yes".
Enable DHCP Option 42	If set to "Yes", the UCM61xx is allowed to get provisioned for NTP Server from DHCP Option 42 in the local server automatically. This will override the manually configured NTP Server. The default setting is "Yes".
Time Zone	Select the proper time zone option so the UCM61xx can display correct time accordingly. The default setting is GMT-05:00 (Eastern Time). If "Self-Defined Tome Zone" is selected, please specify the time zone parameters in "Self-Defined Time Zone" field as described in below option.
Self-Defined Time Zone	If "Self-Defined Time Zone" is selected in "Time Zone" option, users will need define their own time zone following the format below. The syntax is: std offset dst [offset], start [/time], end [/time] Default is set to: MTZ+6MDT+5,M4.1.0,M11.1.0 **MTZ+6MDT+5** This indicates a time zone with 6 hours offset and 1 hour ahead for DST, which is U.S central time. If it is positive (+), the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian); If it is negative (-), the local time zone is east. **M4.1.0,M11.1.0** The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec). The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday…). Normally 1, 2, 3, 4 are used. If 5 is used, it means the

	last iteration of the weekday. The 3rd number indicates weekday: 0,1,2,..,6 (for Sun, Mon, Tues, ... ,Sat). Therefore, this example is the DST which starts from the First Sunday of April to the 1st Sunday of November.

PROVISIONING

OVERVIEW

Grandstream SIP Devices can be configured via Web interface as well as via configuration file through TFTP/HTTP/HTTPS download. All Grandstream SIP devices support a proprietary binary format configuration file and XML format configuration file. The UCM61xx provides a Plug and Play mechanism to auto-provision the Grandstream SIP devices in a zero configuration manner by generating XML config file and having the phone to download it. This allows users to finish the installation with ease and start using the SIP devices in a managed way.

To provision a phone, three steps are involved, i.e., discovery, assignment and provisioning. The UCM61xx creates XML config file to the detected/assigned Grandstream device and accomplishes the following configurations on the device after the provisioning:

- An UCM61xx extension will be assigned and registered on the phone.
- SIP-related network settings such as "NAT traversal" and "Use Random Port" are configured on the phone.
- Call settings such as "Dial Plan" and "Auto Answer".
- LDAP client configurations will be set up automatically on the phone to use the default LDAP directory generated in the UCM61xx LDAP server.

This section explains how zero config works on the UCM61xx. The settings for this feature can be accessed via Web GUI->**PBX**->**Basic/Call Routes**->**Zero Config**.

AUTO PROVISIONING

By default, the Zero Config feature is disabled on the UCM61xx for auto provisioning. It can be turned on in "Auto Provision Settings" under Web GUI->**PBX**->**Basic/Call Routes**->**Zero Config**. Three methods of auto provisioning are used.

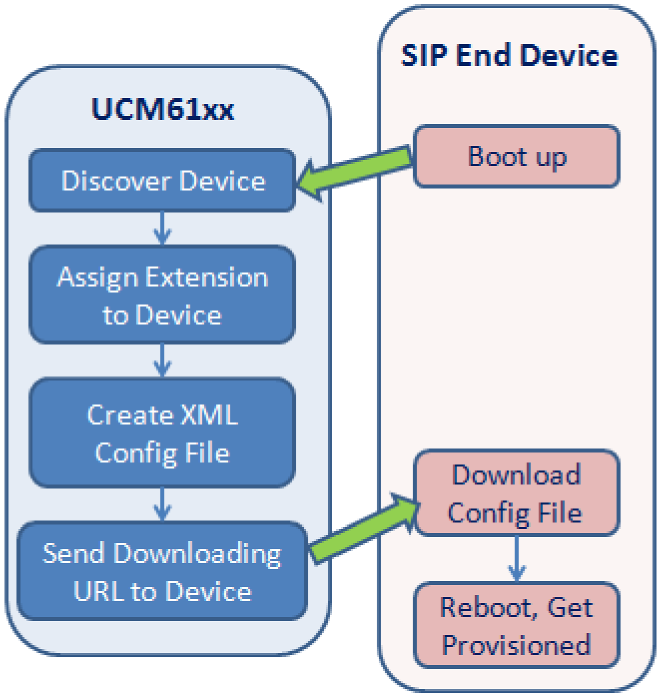


Figure 17: UCM61xx Zero Config

- **SIP SUBSCRIBE**

  When the phone boots up, it sends out SUBSCRIBE to a multicast IP address in the LAN. The UCM61xx discovers it and then sends a NOTIFY with the XML config file URL in the message body. The phone will then use the path to download the config file generated in the UCM61xx and reboot again to take the new configuration.

- **DHCP OPTION 66**

  This method should be used on the UCM6102 because only the UCM6102 has WAN and LAN port with LAN port supporting the router function. When the phone restarts (by default DHCP Option 66 is turned on), it will send out a DHCP DISCOVER request. The UCM6102 receives it and returns DHCP OFFER with the config server path URL in Option 66, for example, http://192.168.2.1:8089/zccgi/. The phone will then use the path to download the config file generated in the UCM61xx.

- **mDNS**

  When the phone boots up, it sends out mDNS query to get the TFTP server address. The UCM61xx will respond with its own address. The phone will then send TFTP request to download the XML config file from the UCM61xx.

To start the auto provisioning process, under Web GUI->**PBX**->**Basic/Call Routes**->**Zero Config**, click on "Auto Provision Settings" and fill in the auto provision information.

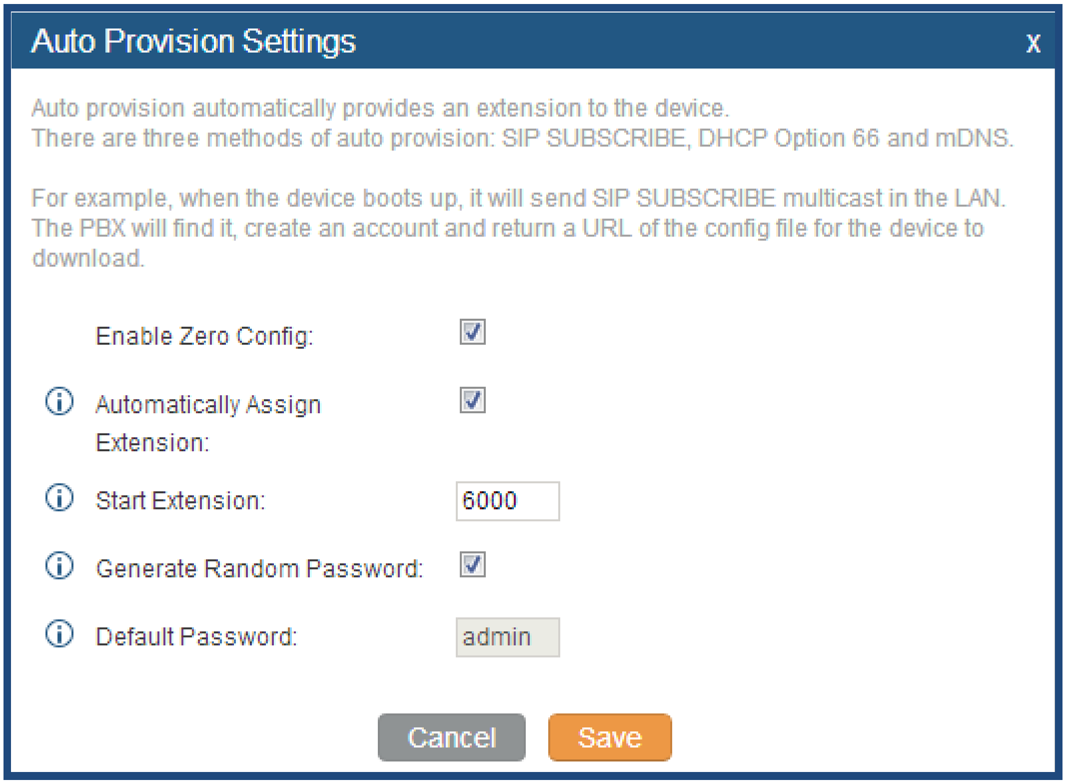


Figure 18: Auto Provision Settings

Table 19: Auto Provision Settings

Enable Zero Config	Enable or disable the zero config feature on the PBX. The default setting is disabled.
Automatically Assign Extension	If enabled, when the device is discovered, the PBX will automatically assign an extension to the device. The default setting is disabled.
Starting Extension	Specify the starting extension to be created/assigned. If the extension is assigned to existing device already, this extension will be skipped and the next available extension will be used. The default setting is 6000.
Generate Random Password	If enabled, random password will be generated for the extension when it's created. Otherwise, default password will be used.
Default Password	Specify default password for the extension if no random password is generated. The default setting is "admin".

Click on "Save" and then reboot the phones to have the discovery and provisioning process started.

MANUAL PROVISIONING

DISCOVERY

Users could manually discover the device by specifying the IP address or scanning the entire network. Three methods are supported to scan the devices.

- PING
- ARP
- SIP MESSAGE (OPTIONS)

Click on "Auto Discover", fill in the scan method and scan IP. The IP address segment will be automatically filled in based on the network mask detected on the UCM61xx. If users need scan the entire network segment, enter 255 (for example, 192.168.40.255) instead of a specific IP address. Then click on "Save" to start discovering the devices within the same network.

Figure 19: Auto Discover

The following figure shows a list of discovered phones. The MAC address, IP Address, Extension (if assigned), Version, Vendor, Model, Connect Status, Create Config, Options (Edit/Delete) are displayed in the list.

1.	000B823E1D8D	192.168.40.249	--	1.0.2.12	Grandstream	GXP2200	Connected	No	
2.	000B823E1D7C	192.168.40.122	--	1.0.1.40	Grandstream	GXP2200	Connected	No	
3.	000B823E5E88	192.168.40.207	--	1.0.5.23	Grandstream	GXP2124	Connected	No	
4.	000B823E175D	192.168.40.145	--	1.0.1.40	Grandstream	GXP2200	Connected	No	
5.	000B823E1D7F	192.168.40.163	--	1.0.2.12	Grandstream	GXP2200	Connected	No	

Figure 20: Discovered Devices

ASSIGNMENT

In the discovered list, click on to open the edit dialog to assign an extension to this device.

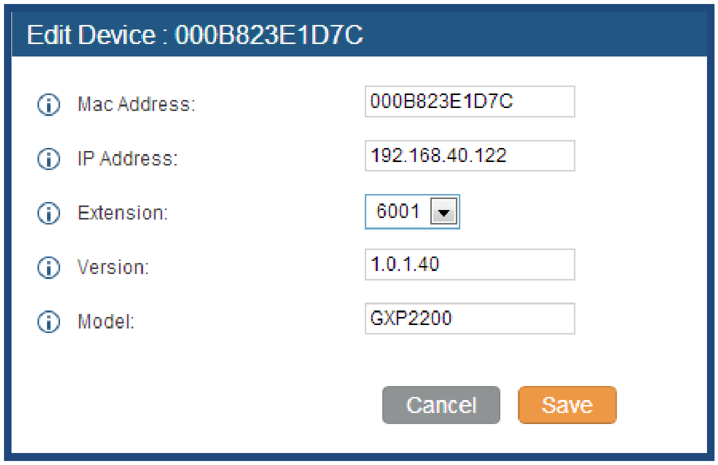


Figure 21: Assign Extension To Device

After saving the edit dialog, the XML config file will be generated in the UCM61xx. Reboot the phone to trigger the phone to download the config file.

CREATE NEW DEVICE

Users could also directly create a new device and assign the extension before the device is discovered by the UCM61xx. Once the device is plugged in, it can then be discovered and get provisioned by the UCM61xx.

Click on "Create New Device" and the following dialog will show. Fill in the MAC address or IP address, and then select the extension to assign to the device. Click on "Save" to add the device to the provision list.

Figure 22: Create New Device

PROVISIONING

After the discovery and assignment, reboot the device. It will download the config file and get provisioned with the assigned extension registered.

EXAMPLES

Depending on the topology, the discovery and provisioning can be done in different ways.

Example 1:

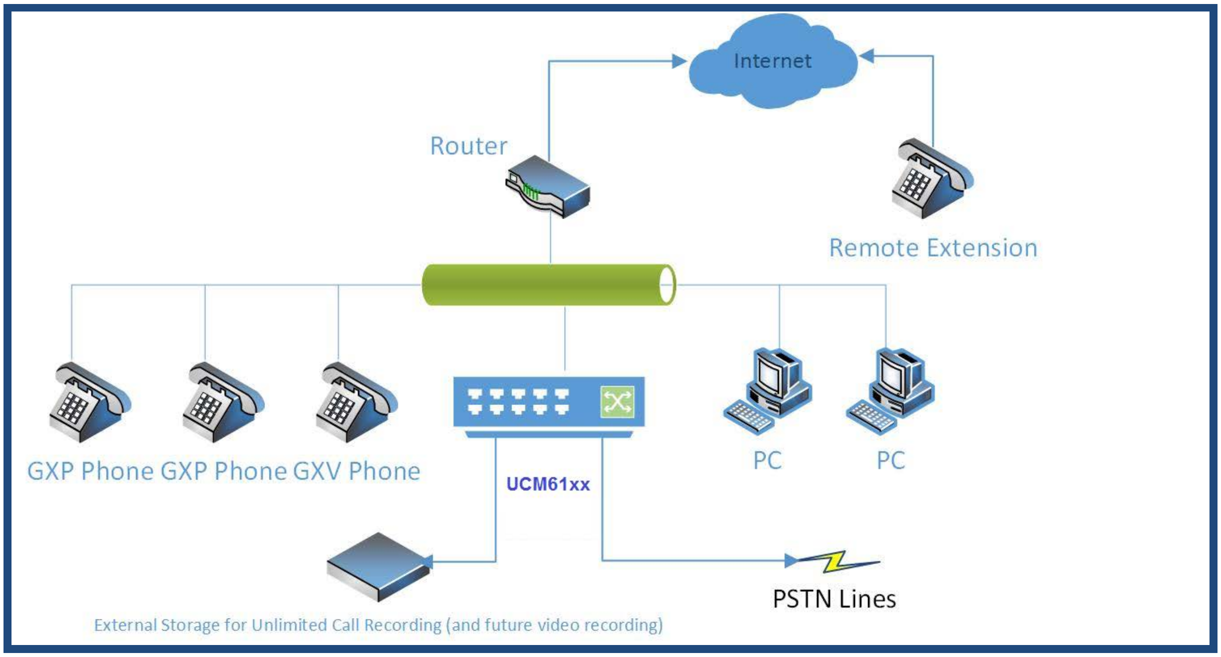


Figure 23: Provisioning Example 1

The above figure shows a common setup among small businesses, where the UCM61xx is placed behind a company's router or firewall. The phones are in the same network as the UCM61xx and can be discovered automatically by UCM61xx using the Zero Config feature.

Example 2:

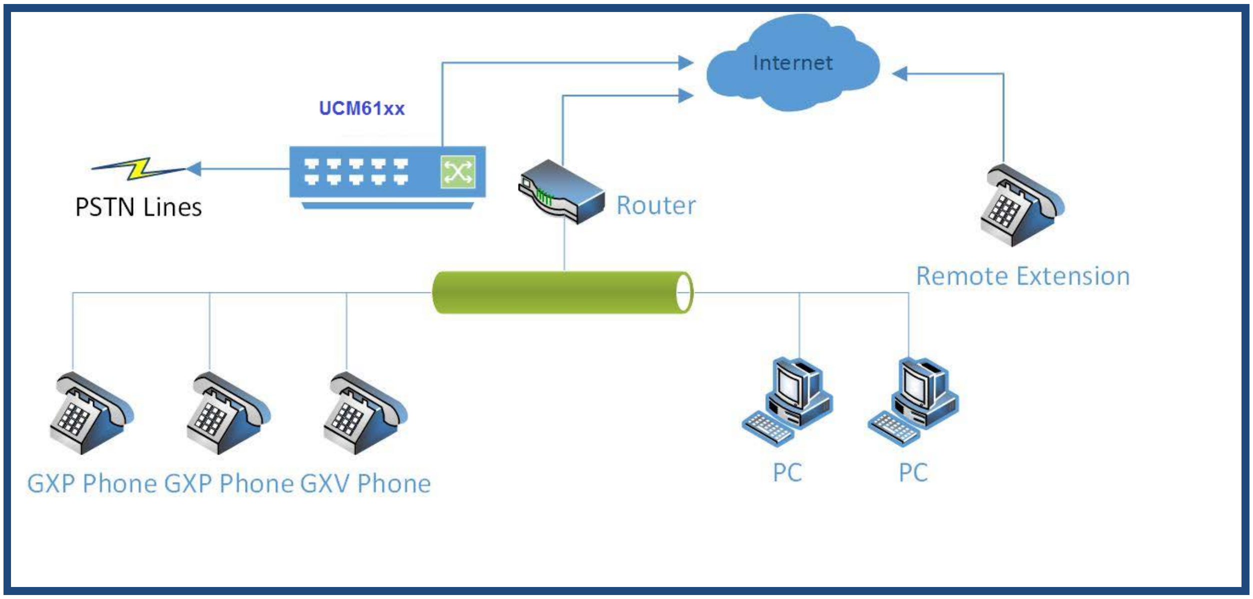


Figure 24: Provisioning Example 2

This is another typical setup. In this setup, the UCM61xx is placed directly over the internet (outside from the network where the phones are deployed). Under this topology, the UCM61xx cannot reach the phones on its own and the typical auto discovery will not work.

In this case, the phones can still be provisioned. But the UCM61xx will need help to get the phones to point itself to the UCM61xx first. One possible solution could be as follows.

- Turn on DHCP Option 66 in the network where the phones are deployed and set the value as: ***option tftp-server-name "http(s)://ucm_ip_address:port/zccgi"***.
- All Grandstream phones have DHCP Option 66 turned on by default.
- Once the phone is provisioned with the DHCP Option 66, it will be redirected to the UCM61xx and send request for the XML config file.
- When the phone requests cfgMAC.xml from the UCM61xx, the UCM61xx will add the phone to the provision list.

EXTENSIONS

CREATE NEW USER

To manually create new user, go to Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**. Click on "Create New User" and a new dialog window will show for users to fill in the extension information. The configuration parameters are as follows.

Table 20: Extension Configuration Parameters

General	
Extension	The extension number associated with the user.
CallerID Name	Configure the CallerID Name associated with the user. Number, letter, or space are allowed.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. **Note:** The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than a outbound rule's privilege in order to make outbound calls from this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purpose.
Enable Voicemail	Enable Voicemail for the user. The default setting is "Yes".
Voicemail Password	Configure Voicemail password (digits only). A random numeric password is automatically generated. It is recommended to use the random generated password for security purpose.
Email Address	Fill in the Email address for the user.
Call Forward Unconditional	Configure the Call Forward Unconditional target number. If not configured, the Call Forward Unconditional feature is deactivated.
Call Forward No Answer	Configure the Call Forward No Answer target number. If not configured,

	the Call Forward No Answer feature is deactivated.
Call Forward Busy	Configure the Call Forward Busy target number. If not configured, the Call Forward Busy feature is deactivated.
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled).
Technology	
SIP	Select "SIP" if the user is using SIP or a SIP device.
IAX	Select "IAX" if the user is using IAX or a IAX device.
Analog Station	Select the FXS port if the user is attached on the analog port of the UCM61xx.
SIP Settings	
NAT	Use NAT when the UCM61xx is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports.
Can Reinvite	By default, the UCM61xx will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the UCM61xx to negotiate endpoint-to-endpoint media routing. The default setting is "No".
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Insecure	• Port: Allow peers matching by IP address without matching port number. • Very: Allow peers matching by IP address without matching port number. Also, authentication of incoming INVITE messages is not required. • No: Normal IP-based peers matching and authentication of incoming INVITE. The default setting is "Port".
Enable Keep-alive	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port. The default setting is "Yes".
Keep-alive Frequency	Configure the Keep-alive interval (in seconds) to check if the host is up.

Other Settings	
SRTP	Enable SRTP for the call. The default setting is disabled.
Fax Detection	Enable to detect Fax signal from the user/trunk during the call and send the received Fax to the Email address configured for this extension. If no Email address can be found for the user, send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension. • Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet can be specified. • A Specific IP Address. Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Skip Trunk Auth	If enabled, users will not need enter the "PIN Set" required by the outbound rule to make outbound calls. The default setting is "No".
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263 and H.263p.

BATCH ADD EXTENSIONS

Under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, click on "Batch Add Extensions" to start adding extensions in batch.

Table 21: Batch Add Extension Parameters

General	
Start Extension	Configure the starting extension number of the batch of extensions to be added.

Create Number	Specify the number of extensions to be added.
Permission	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than a outbound rule's privilege in order to make outbound calls from this rule.
Enable Voicemail	Enable Voicemail for the user. The default setting is "Yes".
SIP/IAX Password	Configure the SIP/IAX password for the users. Three options are available to create password for the batch of extensions. • User Random Password. A random secure password will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Voicemail Password	Configure Voicemail password (digits only) for the users. • User Random Password. A random password in digits will be automatically generated. It is recommended to use this password for security purpose. • Use Extension as Password. • Enter a password to be used on all the extensions in the batch.
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled).
Technology	
SIP	Select "SIP" if the users are using SIP or a SIP device.
IAX	Select "IAX" if the users are using IAX or a IAX device.
SIP Settings	
NAT	Use NAT when the PBX is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports.
Can Reinvite	By default, the PBX will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the PBX to negotiate endpoint-to-endpoint media routing. The default setting is "No".

DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Insecure	• Port: Allow peers matching by IP address without matching port number. • Very: Allow peers matching by IP address without matching port number. Also, authentication of incoming INVITE messages is not required. • No: Normal IP-based peers matching and authentication of incoming INVITE. The default setting is "Port".
Enable Keep-alive	If enabled, empty SDP packet will be sent to the SIP server periodically to keep the NAT port. The default setting is "Yes".
Keep-alive Frequency	Configure the number of seconds for the host to be up for Keep-alive.
IAX Settings	
Max Call Numbers	Configure the maximum number of calls allow for each remote IP address.
Require Call Token	If set to "Yes", call token is required. If set to "Auto", it will lock out users who depend on backward compatibility when peer authentication credentials are shared between physical endpoints. The default setting is "Yes".
Other Settings	
SRTP	Enable SRTP for the call. The default setting is "No".
Fax Detection	Enable to detect Fax signal from the user/trunk during the call and send the received Fax to the Email address configured for this extension. If no Email address can be found for the user, send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
Strategy	This option controls how the extension can be used on devices within different types of network. • Allow All Device in any network can register this extension.

	• Local Subnet Only Only the user in specific subnet can register this extension. Up to three subnet can be specified. • A Specific IP Address. Only the device on the specific IP address can register this extension. The default setting is "Allow All".
Skip Trunk Auth	If enabled, users will not need enter the "PIN Set" required by the outbound rule to make outbound calls. The default setting is "No".
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263 and H.263p.

EDIT EXTENSION

All the UCM61xx extensions are listed under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, with SIP status, Extension, CallerID Name, Technology, IP and Port. Each extension has a checkbox for users to select and options "Edit" "Reboot" "Delete".

- **SIP Status**

  Users can see the following icon for each extension to indicate the SIP status.

  Green: Free
  Blue: Ringing
  Yellow: In Use
  Grey: Unavailable

- **Edit single extension**

  Click on to start editing the extension. The configuration options are listed in ***Table 20: Extension Configuration Parameters***.

- **Reboot the user**

Click on to send NOTIFY reboot event to the device with the extension registered. To successfully reboot the user with the extension registered, "Zero Config" needs to be enabled on the UCM61xx web GUI->**PBX**->**Basic/Call Routes**->**Zero Config**->**Auto Provisioning Settings**.

- **Delete single extension**

  Click on to delete the extension. Or select the checkbox of the extension and then click on "Delete Selected Extensions".

- **Modify selected extensions**

  Select the checkbox for the extension(s). Then click on "Modify Selected Extensions" to edit the extensions in a batch. The configuration options are listed in ***Table 21: Batch Add Extension Parameters***.

- **Delete selected extensions**

  Select the checkbox for the extension(s). Then click on "Delete Selected Extensions" to delete the extension(s).

TRUNKS

ANALOG TRUNKS

Go to Web GUI->**PBX**->**Basic/Call Routes**->**Analog Trunks** to add and edit analog trunks.

- Click on "Create New Analog Trunk" to add a new analog trunk.
- Click on to edit the analog trunk.
- Click on to delete the analog trunk.

ANALOG TRUNK CONFIGURATION

The analog trunk options are listed in the table below.

Table 22: Analog Trunk Configuration Parameters

Channels	Select the channel for the analog trunk. • UCM6102: 2 channels • UCM6104: 4 channels • UCM6108: 8 channels • UCM6116: 16 channels
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.
Advanced Options	
Busy Detection	Busy Detection is used to detect far end hangup or for detecting busy signal. The default setting is "ON".
Busy Tone Count	If "Busy Detection" is enabled, users can specify the number of busy tones to be played before hanging up. The default setting is 2. Better results might be achieved if set to 4, 6 or even 8. Please note that the higher the number, the more time is needed to hangup the channel. However, this might lower the probability to get random hangup.
Congestion Detection	Congestion detection is used to detect far end congestion signal. The default setting is "ON".
Congestion Count	If "Congestion Detection" is enabled, users can specify the number of congestion tones to wait for. The default setting is 2.

Enable Polarity Reversal	If enabled, a polarity reversal will be marked as received when an outgoing call is answered by the remote party. For some countries, a polarity reversal is used for signaling the disconnection of a phone line and the call will be considered as "hangup" on a polarity reversal. The default setting is "No".
Polarity on Answer Delay	When FXO port answers the call, FXS may send a Polarity Reversal. If this interval is shorter than the value of "Polarity on Answer Delay", the Polarity Reversal will be ignored. Otherwise, the FXO will onhook to disconnect the call. The default setting is 600ms.
Current Disconnect Threshold (ms)	This is the periodic time (in ms) that the UCM61xx will use to check on a voltage drop in the line. The default setting is 200ms.
Ring Timeout	Configure the ring timeout (in ms). Trunk (FXO) devices must have a timeout to determine if there was a hangup before the line is answered. This value can be used to configure how long it takes before the UCM61xx considers a non-ringing line with hangup activity.
RX Gain	Configure the RX gain for the receiving channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
TX Gain	Configure the TX gain for the transmitting channel of analog FXO port. The valid range is from -13.5 (dB) to + 12.0 (dB). The default setting is 0.
Use CallerID	Configure to enable CallerID detection. The default setting is "Yes".
Fax Detection	Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
Caller ID Scheme	Select the Caller ID scheme for this trunk. The default setting is "Bellcore/Telcordia".
Tone Country	Select the country for tone settings. If "Custom" is selected, users could manually configure the values for Busy Tone and Congestion Tone. The default setting is "United States of America (USA)".
Busy Tone	Syntax: f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country "Custom" to manually configure Busy Tone value.

	Default value: f1=480@-50,f2=620@-50,c=500/500
Congestion Tone	Syntax: f1=val[@level][,f2=val[@level]],c=on1/off1[-on2/off2[-on3/off3]]; Frequencies are in Hz and cadence on and off are in ms. Frequencies Range: [0, 4000) Busy Level Range: (-300, 0) Cadence Range: [0, 16383]. Select Tone Country "Custom" to manually configure Busy Tone value. Default value: f1=480@-50,f2=620@-50,c=250/250
PSTN Detection	Click on "Detect" to detect the busy tone, Polarity Reversal and Current Disconnect by PSTN. Before the detecting, please make sure there are more than one channel configured and working properly. If the detection has busy tone, the "Tone Country" option will be set as "Custom".

PSTN DETECTION

The UCM61xx provides PSTN detection function to help users detect the busy tone, Polarity Reversal and Current Disconnected by PSTN during analog trunk setup. Select the analog trunk under Web GUI->**PBX**->**Basic/Call Routes**->**Analog Trunks** page first. In the dialog window to edit the trunk, click on "Detect" for "PSTN Detection". The following dialog will show for users to perform the detection.

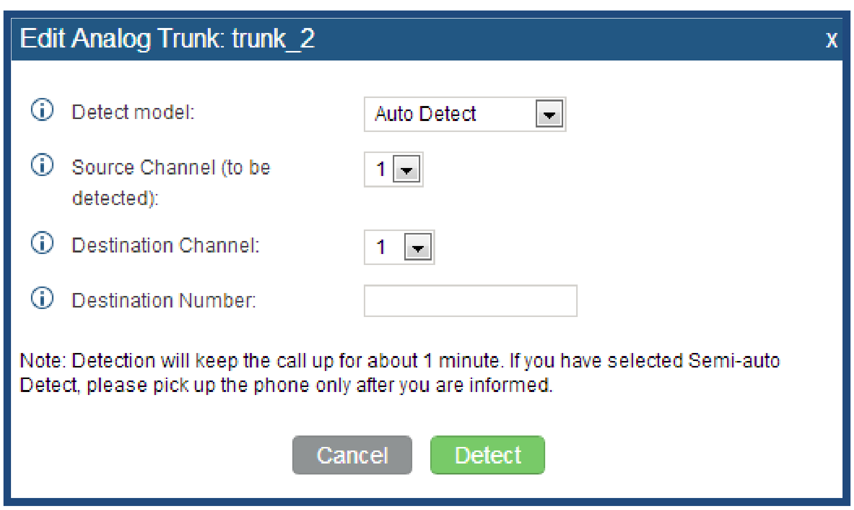


Figure 25: PSTN Detection For Analog Trunk

Table 23: PSTN Detection For Analog Trunk

Detect Model	Select "Auto Detect" or "Semi-auto Detect" for PSTN detection. • Auto Detect Please make sure two or more channels are connected to the UCM61xx and in idle status before starting the detection. During the detection, one channel will be used as caller (Source Channel) and another channel will be used as callee (Destination Channel). The UCM61xx will control the call to be established and hang up between caller and callee to finish the detection. • Semi-auto Detect Semi-auto detection requires answering or hanging up the call manually. Please make sure one channel is connected to the UCM61xx and in idle status before starting the detection. During the detection, source channel will be used as caller and send the call to the configured Destination Number. Users will then need follow the prompts in web GUI to help finish the detection. The default setting is "Auto Detect".
Source Channel	Select the channel to be detected.
Destination Channel	Select the channel to help detect when "Auto Detect" is used.
Destination Number	Configure the number to be called to help detect when "Semi-auto Detect" is used.

Note:

- The PSTN detection process will keep the call up for about 1 minute.
- If "Semi-auto Detect' is used, please pick up the call only after informed from the web GUI prompt.
- Once the detection is successful, the detected parameters "Busy Tone", "Polarity Reversal" and "Current Disconnect by PSTN" will be filled into the corresponding fields in the analog trunk configuration.

VOIP TRUNKS

VoIP trunks can be configured in UCM61xx under Web GUI->**PBX**->**Basic/Call Routes**->**VoIP Trunks**. Once created, the VoIP trunks will be listed with Provider Name, Type, Hostname/IP, Username and Options to edit and detect the trunk.

- Click on "Create New SIP/IAX Trunk" to add a new VoIP trunk.

- Click on to configure detailed parameters for the VoIP trunk.
- Click on to delete the VoIP trunk.

The VoIP trunk options are listed in the table below.

Table 24: VoIP Trunk Configuration Parameters

Create New SIP/IAX Trunk	
Type	Select the VoIP trunk type. • Peer SIP Trunk • Register SIP Trunk • Peer IAX Trunk • Register IAX Trunk
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP providers server of the trunk.
Username	Enter the username to register to the trunk from the provider when "Register SIP Trunk" or "Register IAX Trunk" type is selected.
Password	Enter the password to register to the trunk from the provider when "Register SIP Trunk" or "Register IAX Trunk" type is selected.
Outbound Proxy	Enter the IP address or URL of the outbound proxy for "Register SIP Trunk" type.
Peer SIP Trunk Configuration Parameters	
Provider Name	Configure the provider name for the VoIP trunk. This is a unique label to identify the trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider server of the trunk.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "All - UDP Primary". • UDP Only • TCP Only • TLS Only • All - UDP Primary: UDP is the primary transport protocol when all the other SIP transport methods are available too. • All - TCP Primary: TCP is the primary transport protocol when all the other SIP transport methods are available too.

	• All - TLS Primary: TLS is the primary transport protocol when all the other SIP transport methods are available too.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, H.263p.
Enable Qualify	If enabled, the UCM61xx will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Fax Detection	Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
Register SIP Trunk Configuration Parameters	
Provider Name	Configure the provider name for the VoIP trunk. This is a unique label to identify the trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider server of the trunk.
Transport	Configure the SIP transport protocol to be used in this trunk. The default setting is "All - UDP Primary".

	• UDP Only • TCP Only • TLS Only • All - UDP Primary: UDP is the primary transport protocol when all the other SIP transport methods are available too. • All - TCP Primary: TCP is the primary transport protocol when all the other SIP transport methods are available too. • All - TLS Primary: TLS is the primary transport protocol when all the other SIP transport methods are available too.
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, H.263p.
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the From Header. For example, "trunk.ucm61xx.provider.com" is the From Domain in From Header: sip:1234567@trunk.ucm61xx.provider.com.
From User	Configure the actual user name of the extension. This can be used to override the From Header. There are cases where there is a single ID for registration (single trunk) with multiple DIDs. For example, "1234567" is the From User in From Header: sip:1234567@trunk.ucm61xx.provider.com.
Outbound Proxy Support	Select to enable outbound proxy in this trunk. The default setting is "No".
Outbound Proxy	When outbound proxy support is enabled, enter the IP address or URL of the outbound proxy for "Register SIP Trunk" type.
Enable Qualify	If enabled, the UCM61xx will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Fax Detection	Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
Peer IAX Trunk Configuration Parameters	

Provider Name	Configure the provider name for the VoIP trunk. This is a unique label to identify the trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when the extension has CID configured. The default setting is "No".
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, H.263p.
Enable Qualify	If enabled, the UCM61xx will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Fax Detection	Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
Register IAX Trunk Configuration Parameters	
Provider Name	Configure the provider name for the VoIP trunk. This is a unique label to identify the trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider server of the trunk.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by extension's CID when

	the extension has CID configured. The default setting is "No".
Caller ID	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored. When making outgoing calls, the following rules are used to determine which CallerID will be used if they exist: • The CallerID configured for the extension will be looked up first. • If no CallerID configured for the extension, the CallerID configured for the trunk will be used. • If the above two are missing, the "Global Outbound CID" defined in Web GUI->**PBX**->**Internal Options**->**General** will be used.
CallerID Name	Configure the name of the caller to be displayed when the extension has no CallerID Name configured.
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, H.263p.
Enable Qualify	If enabled, the UCM61xx will regularly send SIP OPTIONS to the device to check if the device is still online. The default setting is "No".
Fax Detection	Enable to detect Fax signal from the trunk during the call and send the received Fax to the default Email address in Fax setting page under web GUI->PBX->Internal Options->Fax/T.38. **Note:** If enabled, Fax Pass-through cannot be used.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".

CALL ROUTES

OUTBOUND ROUTES

In the UCM61xx, an outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks (e.g., "Local" 7-digit dials through a FXO while "Long distance" 10-digit dials through a low-cost SIP trunk). Users can also set up a failover trunk to be used when the primary trunk fails.

Go to Web GUI->**PBX**->**Basic/Call Routes**->**Outbound Routes** to add and edit outbound rules.

- Click on "Create New Outbound Rule" to add a new outbound route.
- Click on to edit the outbound route.
- Click on to delete the outbound route.
- Click on to move the outbound route up/down to arrange the priority of the outbound rule.

  The outbound rule listed on the top has higher priority. When the dialing pattern matches two or more outbound rules (for example, the same pattern is configured for 2 different trunks; or dialing out 1000 matches pattern 1xxx for trunk 1 and pattern 100x for trunk 2), the one list on the top will be used.

Table 25: Outbound Route Configuration Parameters

Calling Rule Name	Configure the name of the calling rule (e.g., local, long_distance, and etc). Letters, digits, _ and - are alllowed.
Pattern	• All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9]: Any digit from 1 to 9.
Privilege Level	Select privilege level for the outbound rule. • Internal: The lowest level required. All users can use this rule. • Local: Users with Local, National, or International level are allowed

	to use this rule. • National: Users with National or International level are allowed to use this rule. • International: The highest level required. Only users with international level can use this rule.
Pin Set	Configure the password for users to use this rule when making outbound calls.
Send This Call Trough Trunk	
Use Trunk	Select the trunk for this outbound rule.
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. Example: The users will dial 9 as the first digit of a long distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Use Failover Trunk	
Failover Trunk	Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down. If "Use Failover Trunk" is enabled and "Failover trunk" is defined, the calls that cannot be placed via the regular trunk may have a secondary trunk to go through. Example: The user's primary trunk is a VoIP trunk and the user would like to use the PSTN when the VoIP trunk is not available. The PSTN trunk can be configured as the failover trunk of the VoIP trunk.
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. Example: The users will dial 9 as the first digit of a long distance calls. However, 9 should not be sent out via analog lines and the PSTN line. In this case, 1 digit should be stripped before the call is placed.
Prepend	Specify the digits to be prepended before the call is placed via the trunk.

	Those digits will be prepended after the dialing number is stripped.

INBOUND ROUTES

Inbound routes can be configured via Web GUI->**PBX**->**Basic/Call Routes**->**Inbound Routes**.

- Click on "Create New Inbound Rule" to add a new inbound route.
- Click on "DID Features" to configure DID features for the inbound route.
- Click on "Blacklist" do configure blacklist.
- Click on to edit the inbound route.
- Click on to delete the inbound route.

INBOUND RULE CONFIGURATIONS

Table 26: Inbound Rule Configuration Parameters

Trunks	Select the trunk to configure the inbound rule.
DID Pattern	• All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9]: Any digit from 1 to 9.
Privilege Level	Select privilege level for the inbound rule. • Internal: The lowest level required. All users can use this rule. • Local: Users with Local, National or International level are allowed to use this rule. • National: Users with National or International level are allowed to use this rule. • International: The highest level required. Only users with international level can use this rule.
Default Destination	Select the default destination for the inbound call.

	• Extension • Extension's voicemail • Call Queue • Ring Group • Voicemail Access Code • Fax • Operator • Hangup • Congestion • By DID
Time Condition	
Start Time	Select the start time "hour:minute" for the trunk to use the inbound rule.
End Time	Select the end time "hour:minute" for the trunk to use the inbound rule.
Date	Select "By Week" or "By Day" and specify the date for the trunk to use the inbound rule.
Week	Select the day in the week to use the inbound rule.
Destination	Select the default destination for the inbound call. • Extension • Extension's voicemail • Call Queue • Ring Group • Voicemail Access Code • Fax • Operator • Hangup • Congestion • By DID
DID Features	
Dial Trunk	If enabled, external users can dial outbound calls by DID through inbound trunks.
DID Destination	Select the DID destination. Only the selected category can be reached by DID. • User Extension. This is selected by default • Conference • Call Queue • Ring Group • Page/Intercom Group

BLACKLIST CONFIGURATIONS

In the UCM61xx, Blacklist is supported in all inbound routes. Users could enable the Blacklist feature, manage the Blacklist by clicking on "Blacklist".

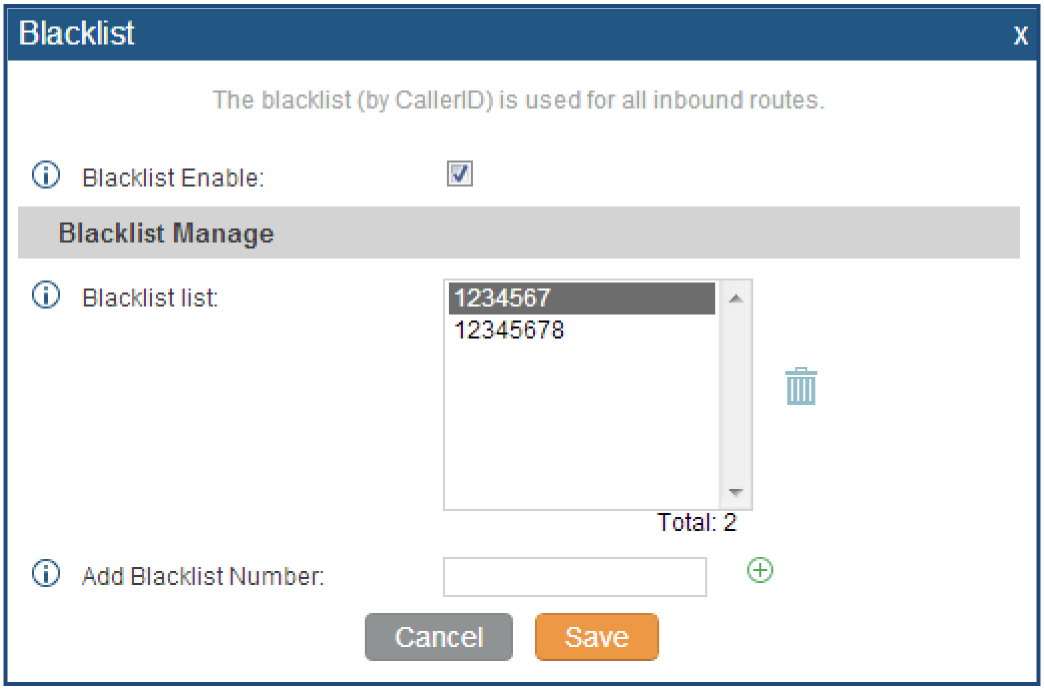


Figure 26: Blacklist Configuration Parameters

- Select the checkbox for "Blacklist Enable" to turn on Blacklist feature for all inbound routes. Blacklist is disabled by default.
- Enter a number in "Add Blacklist Number" field and then click to add to the list.
- To remove a number from the Blacklist, select the number in "Blacklist list" and click on .

Note:

Users could also add a number to the Blacklist or remove a number from the Blacklist by dialing the feature code for "Blacklist Add' and "Blacklist Remove" from an extension. The feature code can be configured under Web GUI->**PBX**->**Internal Options**->**Feature Codes**.

CONFERENCE BRIDGE

The UCM61xx supports conference bridge allowing multiple bridges used at the same time:

- UCM6102/6104 supports up to 3 conference bridges allowing up to 25 simultaneous PSTN or IP participants.
- UCM6108/6116 supports up to 6 conference bridges allowing up to 32 simultaneous PSTN or IP participants.

The conference bridge configurations can be accessed under Web GUI->**PBX**->**Call Features**->**Conference**. In this page, users could create, edit, view, invite, manage the participants and delete conference bridges. The conference bridge status and conference call recordings (if recording is enabled) will be displayed in this web page as well.

CONFERENCE BRIDGE CONFIGURATIONS

- Click on "Create New Conference Room" to add a new conference bridge.
- Click on to edit the conference bridge.
- Click on to delete the conference bridge.

Table 27: Conference Bridge Configuration Parameters

Extension	Configure the conference number for the users to dial into the conference.
Password	When configured, the users who would like to join the conference call must enter this password before accessing the conference bridge. **Note:** If "Public Mode" is enabled, the password is not required to join the conference bridge thus this field is invalid.
Admin Password	Configure the password to join the conference bridge as administrator. Conference administrator can manage the conference call via IVR (if "Enable Caller Menu" is enabled) as well as invite other parties to join the conference by dialing "0" (permission required from the invited party) or "1" (permission not required from the invited party) during the conference call.

	Note: If "Public Mode" is enabled, the password is not required to join the conference bridge thus this field is invalid.
Enable Caller Menu	If enabled, conference participant could press the * key to access the conference bridge menu. The default setting is "No".
Record Conference	If enabled, the calls in this conference bridge will be recorded automatically in a .wav format file. All the recording files will be displayed and can be downloaded in the conference web page. The default setting is "No".
Quiet Mode	If enabled, if there are users joining or leaving the conference, voice prompt or notification tone won't be played. The default setting is "No". **Note:** "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.
Wait For Admin	If enabled, the participants will not hear each other until the conference administrator joins the conference. The default setting is "No". **Note:** If "Quiet Mode" is enabled, the voice prompt for "Wait For Admin" will not be announced.
Enable User Invite	If enabled, users could press 0 to invite other users (with the users' permission) or press 1 to invite other users (without the user's permission) to join the conference. The default setting is "No". **Note:** Conference administrator can always invite other users without enabling this option.
Announce Callers	If enabled, the caller will be announced to all conference participants when there the caller joins the conference. The default setting is "No". **Note:** "Quiet Mode" and "Announce Callers" cannot be enabled at the same time.
Enable Jitter Buffer	If enabled, the voice quality for conference call will be improved. However, this could cause voice delay and increase system resource usage. The default setting is "No".
Public Mode	If enabled, no authentication will be required when joining the

	conference call. The default setting is "Yes".
Play Hold Music For First Caller	If enabled, the UCM61xx will play Hold music to the first participant in the conference until another user joins in. The default setting is "No".
Skip Authentication When Invite User via Trunk from Web GUI	If enabled, the invitation from Web GUI for a conference bridge with password will skip the authentication for the invited users. The default setting is "No".

JOIN A CONFERENCE CALL

Users could dial the conference bridge extension to join the conference. If password is required, enter the password to join the conference as a normal user, or enter the admin password to join the conference as administrator.

INVITE OTHER PARTIES TO JOIN CONFERENCE

When using the UCM61xx conference bridge, there are two ways to invite other parties to join the conference.

- Invite from Web GUI.

For each conference bridge in UCM61xx Web GUI->**PBX**->**Call Features**->**Conference**, there is an icon  for option "Invite a participant". Click on it and enter the number of the party you would like to invite. Then click on "Add". A call will be sent to this number to join it into the conference.

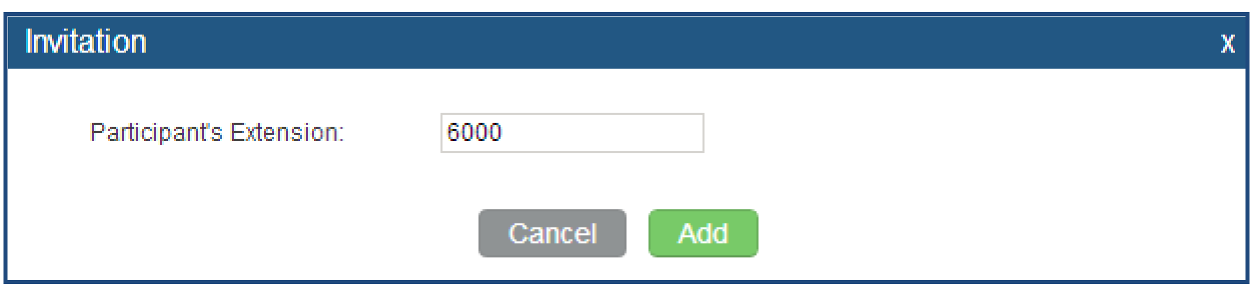


Figure 27: Conference Invitation From Web GUI

Note:
When a user invite other parties to join a conference from Web GUI, the user doesn't have to be in the conference bridge.

- Invite by dialing 0 or 1 during conference call.

A conference participant can invite other parties to the conference by dialing from the phone during the conference call. Please make sure option "Enable User Invite" is turned on for the conference bridge first. Enter 0 or 1 during the conference call. Follow the voice prompt to input the number of the party you would like to invite. A call will be sent to this number to join it into the conference.

0: If 0 is entered to invite other party, once the invited party picks up the invitation call, a permission will be asked to "accept" or "reject" the invitation before joining the conference.
1: If 1 is entered to invite other party, no permission will be required from the invited party.

Note:
Conference administrator can always invite other parties from the phone during the call by entering 0 or 1. To join a conference bridge as administrator, enter the admin password when joining the conference. A conference bridge can have multiple administrators.

DURING THE CONFERENCE

During the conference call, users can manage the conference from web GUI or IVR.

- Manage the conference call from Web GUI.

  Log in UCM61xx web GUI during the conference call, the participants in each conference bridge will be listed.

  1. Click on to kick a participant from the conference.
  2. Click on to mute the participant.
  3. Click on to lock this conference bridge so that other users cannot join it anymore.
  4. Click on to invite other users into the conference bridge.

Note:
When there is participant in the conference, the conference bridge configuration cannot be modified.

- Manage the conference call from IVR.

  If "Enable Caller Menu" is enabled, conference participant can input * to enter the IVR menu for the conference. Please see options listed in the table below.

Table 28: Conference Caller IVR Menu

Conference Administrator IVR Menu	
1	Mute/unmute yourself.
2	Lock/unlock the conference bridge.
3	Kick the last joined user from the conference.
4	Decrease the volume of the conference call.
5	Decrease your volume .
6	Increase the volume of the conference call.
7	Increase your volume.
8	More options. • 1: List all users currently in the conference call. • 2: Kick all non-Administrator participants from the conference call. • 3: Mute/Unmute all non-Administrator participants from the conference call. • 4: Record the conference call. • 8: Exit the caller menu and return to the conference.
Conference User IVR Menu	
1	Mute/unmute yourself.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.
7	Increase your volume.
8	Exit the caller menu and return to the conference.

RECORD CONFERENCE

The UCM61xx allows users to record the conference call and retrieve the recording from web GUI->**PBX**->**Call Features**->**Conference**.

To record the conference call, when the conference bridge is in idle, enable "Record Conference" from the conference bridge configuration dialog. Save the setting and apply the change. When the conference call starts, the call will be automatically recorded in .wav format.

The recording files will be listed as below once available. Users could click on to download the recording or click on to delete the recording.

Name	Room	Date	Size	Options
meetme-conf-rec-6300-1372865271.25.wav	6300	2013-07-03 12:39:38 UTC-03:00	10.61 MB	
meetme-conf-rec-6300-1372451238.6.wav	6300	2013-06-28 17:27:46 UTC-03:00	120.04 KB	
meetme-conf-rec-6300-1372205127.347.wav	6300	2013-06-25 21:05:56 UTC-03:00	82.86 KB	
meetme-conf-rec-6300-1372867161.40.wav	6300	2013-07-03 13:10:29 UTC-03:00	10.17 MB	
meetme-conf-rec-6300-1372864546.12.wav	6300	2013-07-03 12:16:01 UTC-03:00	35.67 KB	
meetme-conf-rec-6300-1372866438.36.wav	6300	2013-07-03 12:47:47 UTC-03:00	322.86 KB	
meetme-conf-rec-6300-1372204987.337.wav	6300	2013-06-25 21:03:30 UTC-03:00	315.98 KB	
meetme-conf-rec-6300-1372864583.17.wav	6300	2013-07-03 12:16:36 UTC-03:00	65.67 KB	
meetme-conf-rec-6300-1370385024.71.wav	6300	2013-06-04 19:35:28 UTC-03:00	4.22 MB	

Figure 28: Conference Recording

IVR

CONFIGURE IVR

IVR configurations can be accessed under the UCM61xx Web GUI->**PBX**->**Call Features**->**IVR**. Users could create, edit, view and delete an IVR.

- Click on "Create New IVR" to add a new IVR.
- Click on to edit the IVR configuration.
- Click on to delete the IVR.

Table 29: IVR Configuration Parameters

Name	Configure the name of the IVR. Letters, digits, _ and - are allowed.
Extension	Enter the extension number for users to access the IVR.
Dial Other Extensions	If enabled, all callers to the IVR can dial other extensions. The default setting is "No".
Dial Trunk	If enabled, all callers to the IVR is allowed to use trunk. The permission must be configured for the users to use the trunk first. The default setting is "No".
Permission	Assign permission level for outbound calls. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". **Note:** Users need to have the same level as or higher level than a outbound rule's privilege in order to make outbound calls from this rule.
Welcome Prompt	Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add additional audio file under web GUI->**Internal Options**->**IVR Prompt**.
Timeout	After playing the prompts in the IVR, the UCM61xx will wait for the DTMF entry within the timeout (in seconds). If no DTMF entry is detected within the timeout, a timeout prompt will be played. The default setting is 10 seconds.
Timeout Prompt	Select the prompt message to be played when timeout occurs.
Invalid Prompt	Select the prompt message to be played when an invalid extension is

	pressed.
Timeout Repeat Loops	Configure the number of times to repeat the prompt if no DTMF input is detected. When the loop ends, it will go to the timeout destination if configured, or hang up. The default setting is 4.
Invalid Repeat Loops	Configure the number of times to repeat the prompt if the DTMF input is invalid. When the loop ends, it will go to the invalid destination if configured, or hang up. The default setting is 4.
Key Press Event	Select the event for each key pressing for 0-9, *, Timeout and Invalid. The event options are: • Extension • VoiceMail • Conference Rooms • VoiceMail Group • IVR • Ring Group • Queues • Page Group • IVR Prompt • Hangup

CREATE IVR PROMPT

To record new IVR prompt or upload IVR prompt to be used in IVR, click on "Prompt" next to the "Welcome Prompt" option and the users will be redirected to IVR Prompt page. Or users could go to Web GUI->**PBX**->**Internal Options**->**IVR Prompt** page directly.

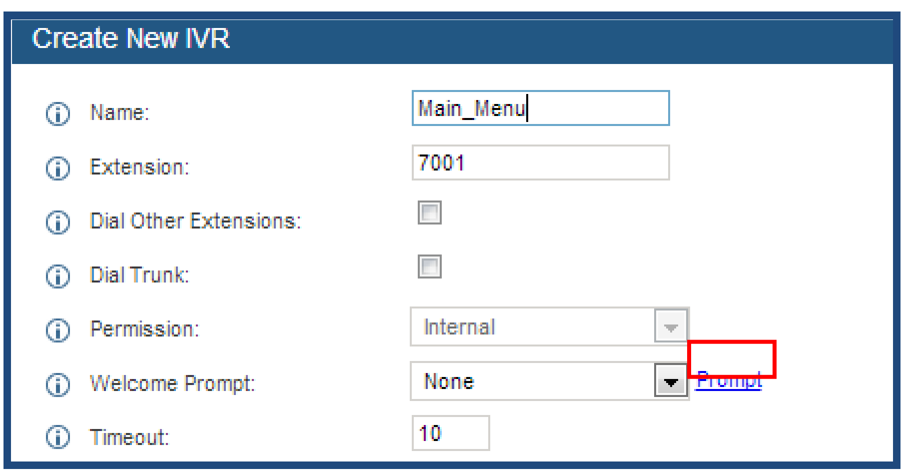


Figure 29: Click On Prompt To Create IVR Prompt

Once the IVR prompt file is successfully added to the UCM61xx, it will be added into the prompt list options for users to select in different IVR scenarios.

RECORD NEW IVR PROMPT

In the UCM61xx Web GUI->**PBX**->**Internal Options**->**IVR Prompt** page, click on "Record New IVR Prompt" and follow the steps below to record new IVR prompt.

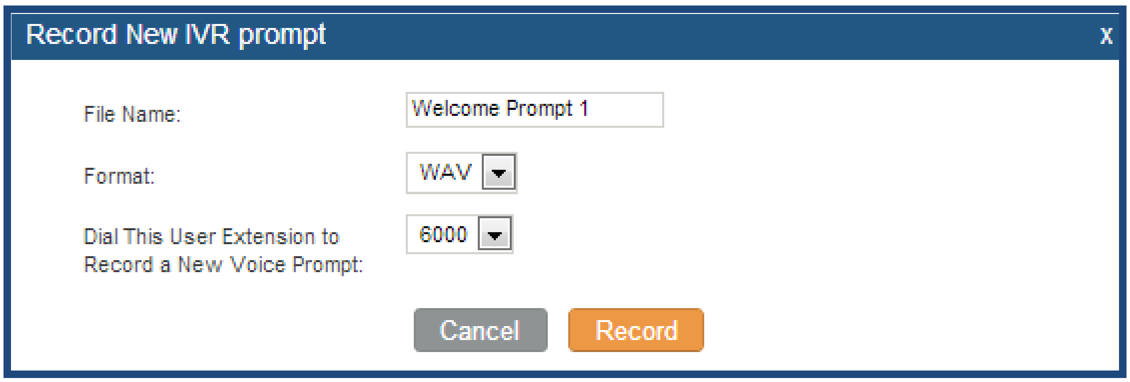


Figure 30: Record New IVR Prompt

- Specify the IVR file name.
- Select the format (GSM or WAV) for the IVR prompt file to be recorded.
- Select the extension to receive the call from the UCM61xx to record the IVR prompt.
- Click the "Record" button. A request will be sent to the UCM61xx. The UCM61xx will then call the extension for recording the IVR prompt from the phone.
- Pick up the call from the extension and start the recording following the voice prompt.
- The recorded file will be listed in the IVR Prompt web page. Users could select to re-record, play or delete the recording.

UPLOAD IVR PROMPT

If the user has a pre-recorded IVR prompt file, click on "Upload IVR Prompt" in Web GUI->**PBX**->**Internal Options**->**IVR Prompt** page to upload the file to the UCM61xx. The following are required for the IVR prompt file to be successfully uploaded and used by the UCM61xx:

- PCM encoded.
- 16 bits.
- 8000Hz mono.
- In .mp3 or .wav format; or raw/ulaw/alaw/gsm file with .ulaw or .alaw suffix.
- File size under 5M.

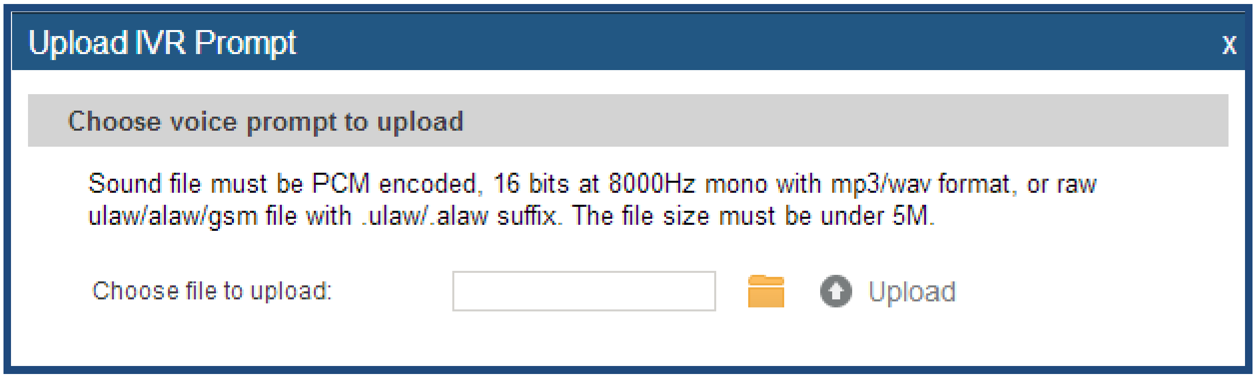


Figure 31: Upload IVR Prompt

Click on to select audio file from local PC and click on to start uploading. Once uploaded, the file will appear in the IVR Prompt web page.

LANGUAGE SETTINGS FOR VOICE PROMPT

The UCM61xx supports multiple languages in web GUI as well as system voice prompt. The following languages are currently supported in system voice prompt:

English
Chinese
German
French
Arabic
Italian
Spanish
Polish
Portuguese
Russian

English and Chinese voice prompts are built in with the UCM61xx already. The other languages provided by Grandstream can be downloaded and installed from the UCM61xx web GUI directly. Additionally, users could customize their own voice prompts, package them and upload to the UCM61xx.

Language settings for voice prompt can be accessed under Web GUI->**PBX**->**Internal Options**->**Language**.

DOWNLOAD AND INSTALL VOICE PROMPT PACKAGE

To download and install voice prompt package in different languages from UCM61xx web GUI, click on "Check Prompt List" button.

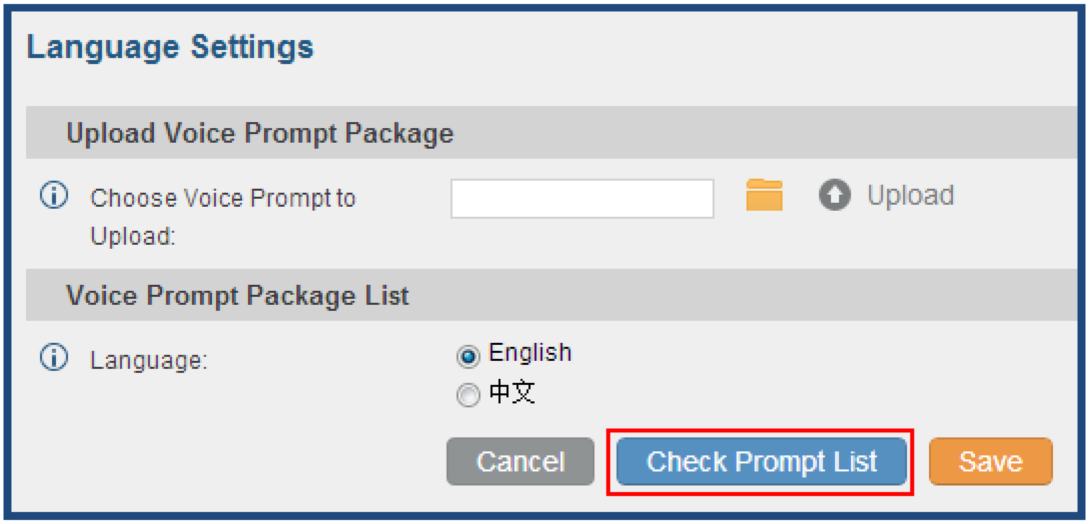
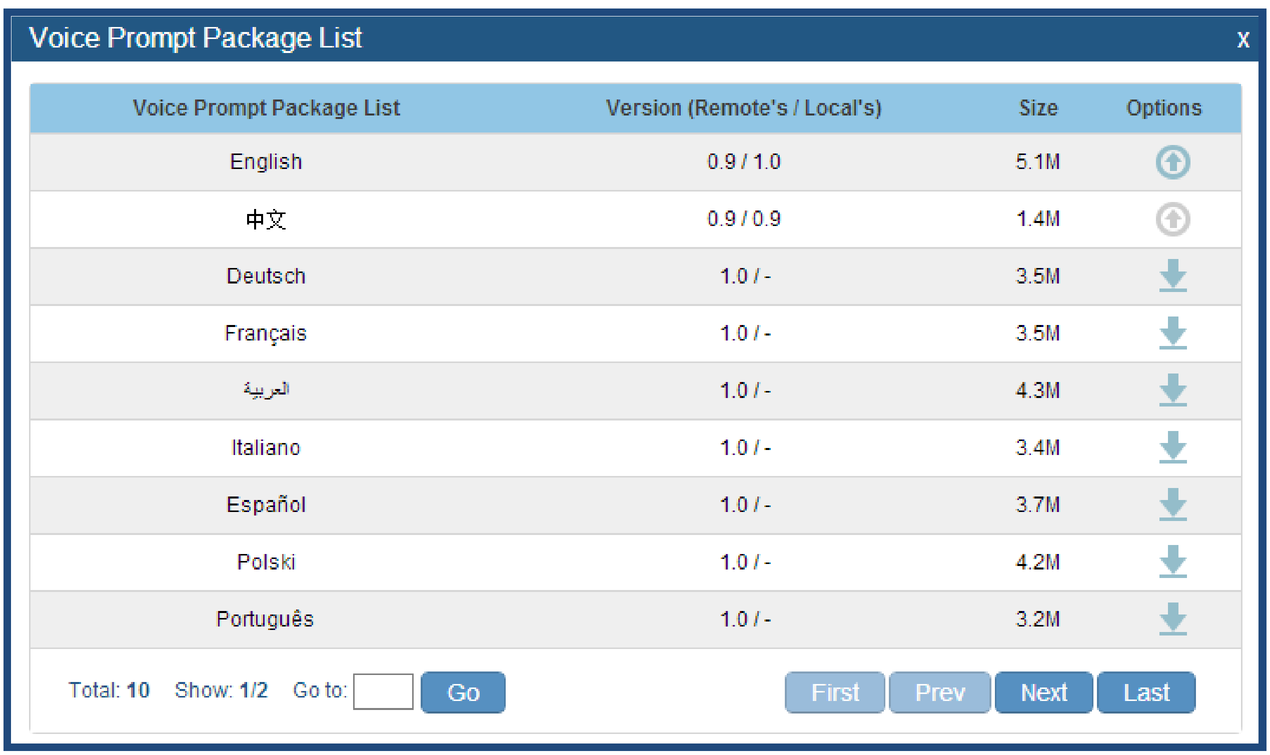


Figure 32: Language Settings For Voice Prompt

A new dialog window of voice prompt package list will be displayed. Users can see the version number (latest version available V.S. current installed version), package size and options to upgrade or download the language.

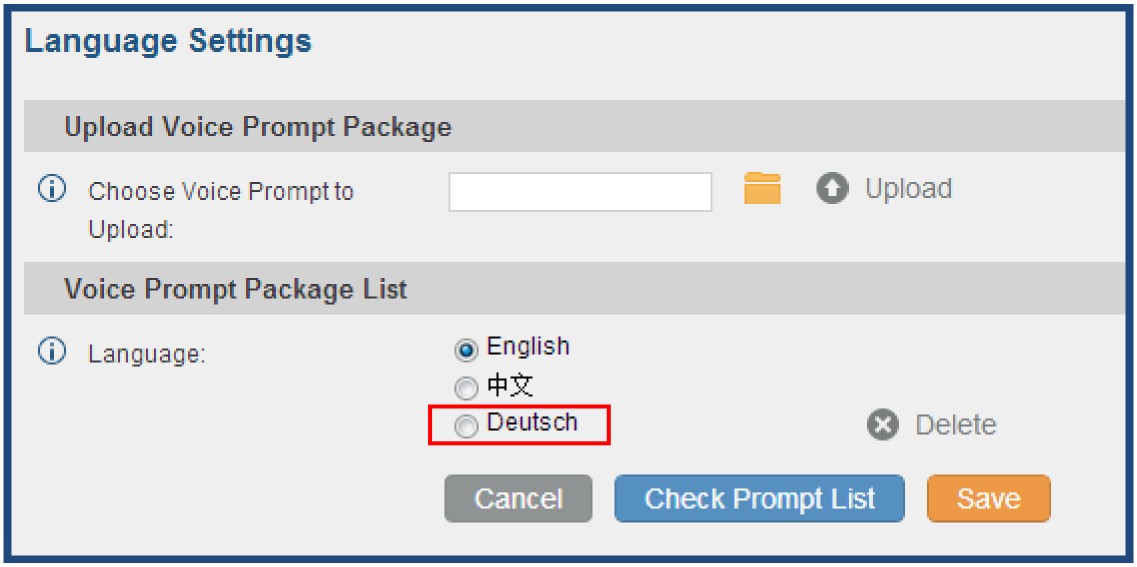

Voice Prompt Package List

Voice Prompt Package List	Version (Remote's / Local's)	Size	Options
English	0.9 / 1.0	5.1M	
中文	0.9 / 0.9	1.4M	
Deutsch	1.0 / -	3.5M	
Français	1.0 / -	3.5M	
العربية	1.0 / -	4.3M	
Italiano	1.0 / -	3.4M	
Español	1.0 / -	3.7M	
Polski	1.0 / -	4.2M	
Português	1.0 / -	3.2M	

Total: 10 Show: 1/2 Go to: Go First Prev Next Last

Figure 33: Voice Prompt Package List

Click on to download the language to the UCM61xx. The installation will be automatically started once the downloading is finished.



A new language option will be displayed after successfully installed. Users then could select it to apply in the UCM61xx system voice prompt or delete it from the UCM61xx.

CUSTOMIZE AND UPLOAD VOICE PROMPT PACKAGE

The UCM61xx provides interface from web GUI for users to customize their own voice prompts. Users could directly upload the package from web GUI. For detailed instructions on voice prompt customizing and uploading, please refer to the link below:

http://www.grandstream.com/products/ucm_series/ucm61xx/documents/ucm61xx_voiceprompt_customization.zip

VOICEMAIL

CONFIGURE VOICEMAIL

If the voicemail is enabled for UCM61xx extensions, the configurations of the voicemail can be globally set up and managed under Web GUI->**PBX**->**Call Features**->**Voicemail**.

Table 30: Voicemail Settings

Max Greeting	Configure the maximum number of seconds for the voicemail greeting. The default setting is 60 seconds.
Dial '0' For Operator	If enabled, the caller can press 0 to exit the voicemail application and connect to the configured operator's extension. The operator extension can be configured under web GUI->**PBX**->**Internal Options**->**General**.
Max Messages Per Folder	Configure the maximum number of messages per folder in users' voicemail. The valid range 10 to 1000. The default setting is 50.
Max Message Time	Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the max message time. The default setting is 15 minutes. The available options are: • 1 minute • 2 minutes • 5 minutes • 15 minutes • 30 minutes • Unlimited
Announce Message Caller-ID	If enabled, the caller ID of the user who has left the message will be announced at the beginning of the voicemail message. The default setting is "No".
Announce Message Duration	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
Play Envelope	If enabled, a brief introduction (received time, received from, and etc) of each message will be played when accessed from the voicemail application. The default setting is "Yes".
Allow Users To Review	If enabled, users can review the message following the IVR before sending the message out. The default setting is "No".

VOICEMAIL EMAIL SETTINGS

The UCM61xx can be configured to send the voicemail as attachment to Email. Click on "Email Settings For Voicemails" button to configure the Email attributes and content.

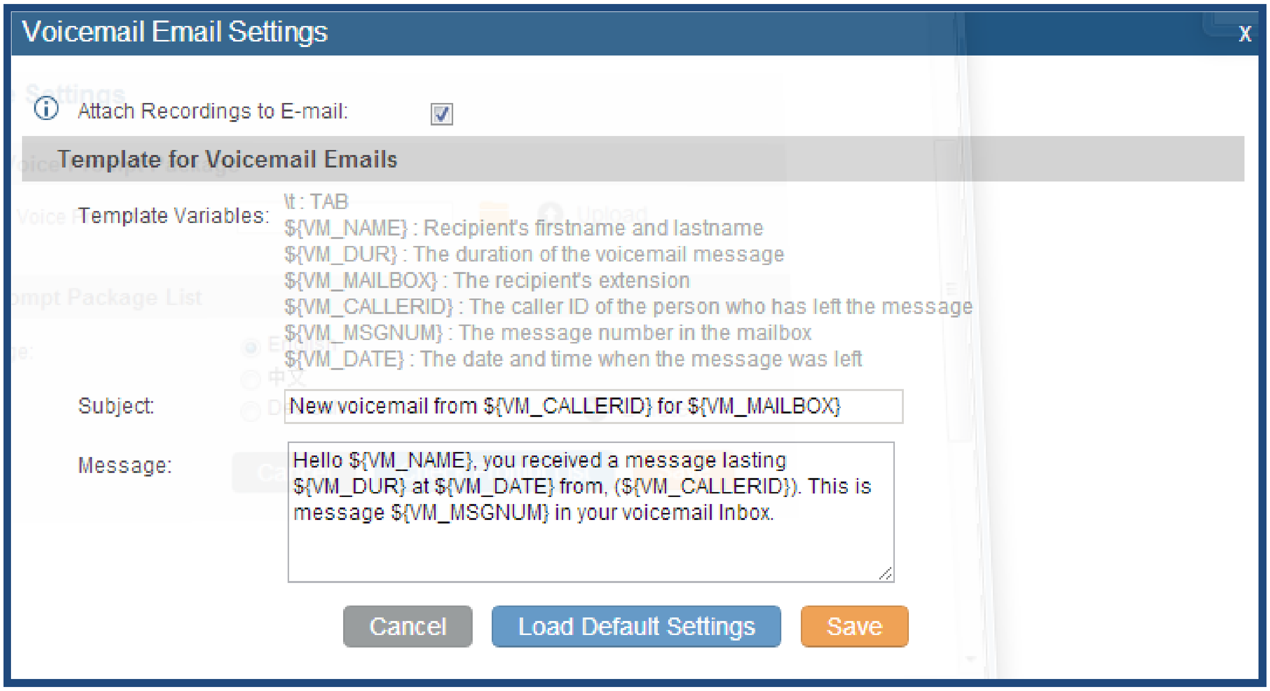


Figure 34: Voicemail Email Settings

Table 31: Voicemail Email Settings

Attach Recordings to E-Mail	If enabled, voicemails will be sent to user's Email address. The default setting is "Yes".
Template For Voicemail Emails	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the users. The template variables are: • \t: TAB • ${VM_NAME}: Recipient's first name and last name • ${VM_DUR}: The duration of the voicemail message • ${VM_MAILBOX}: The recipient's extension • ${VM_CALLERID}: The caller ID of the person who has left the message • ${VM_MSGNUM}: The number of messages in the mailbox • ${VM_DATE}: The date and time when the message is left

Click on "Load Default Settings" button to view the default template as an example.

CONFIGURE VOICEMAIL GROUP

The UCM61xx supports voicemail group and all the extensions added in the group will receive the voicemail to the group extension. The voicemail group can be configured under Web GUI->**PBX**->**Call Features**->**Voicemail Group**. Click on "Create New Voicemail Group" to configure the group.

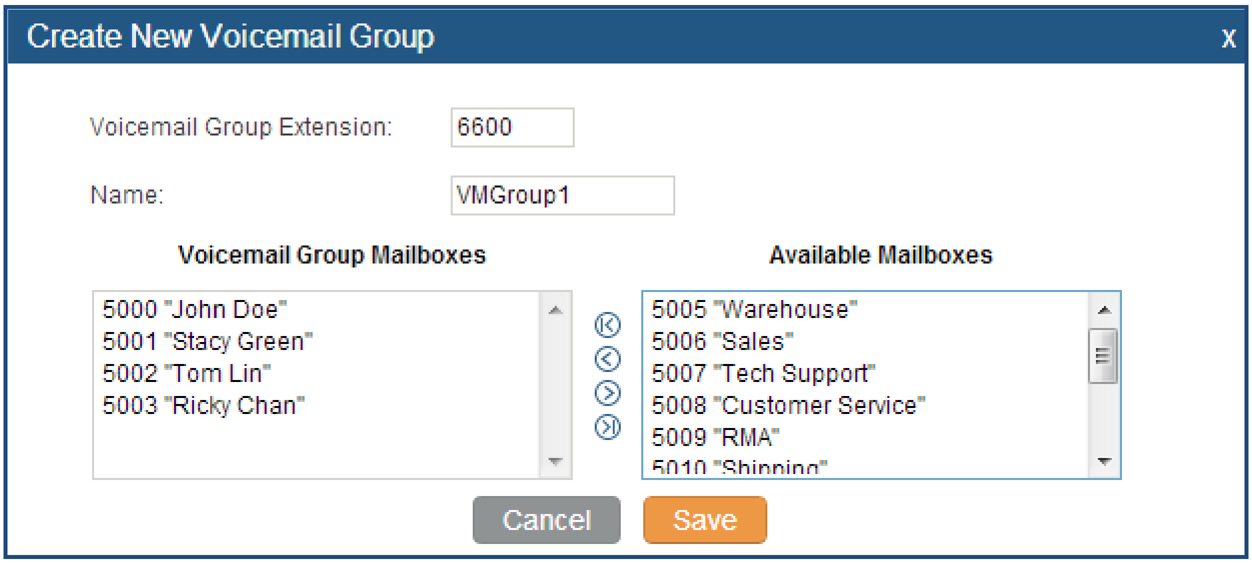


Figure 35: Voicemail Group

- **Voicemail Group Extension**
  Enter the Voicemail Group Extension. The voicemail messages left to this extension will be forwarded to all the voicemail group members.
- **Name**
  Configure the Name to identify the voicemail group. Letters, digits, _ and - are allowed.
- **Voicemail Group Mailboxes**
  Select available mailboxes from the right list and add them to the left list. The extensions need to have voicemail enabled to be listed in available mailboxes list.
- Click on "Save" to finish the configuration.

RING GROUP

The UCM61xx supports ring group feature with different ring strategies applied to the ring group members. This section describes the ring group configuration on the UCM61xx.

CONFIGURE RING GROUP

Ring group settings can be accessed via Web GUI->**PBX**->**Call Features**->**Ring Group**.

Figure 36: Ring Group

- Click on "Create New Ring Group" to add ring group.
- Click on to edit the ring group. The following table shows the ring group configuration parameters.
- Click on to delete the ring group.

Table 32: Ring Group Parameters

Ring Group Name	Configure ring group name to identify the ring group. Letters, digits, _ and - are allowed.
Extension	Configure the ring group extension.
Ring Group Members	Select available users from the right side to the ring group member list on the left side. Click on to arrange the order.
Ring Strategy	Select the ring strategy. • Ring simultaneously. Ring all the members at the same time when there is incoming call to the ring group extension. If any of the member answers the call, it will stop ringing. • Ring in order. Ring the members with the order configured in ring group list. If the first member doesn't answer the call, it will stop ringing the first member and start ringing the second member.

Ring Timeout on Each Member	Configure the number of seconds to ring each member. If set to 0, it will keep ringing The default setting is 30 seconds. **Note:** The actual ring timeout might be overridden by users if the phone has ring timeout settings as well.
Enable Voicemail	If enabled, users could select to use the ring group extension as the voicemail or select another extension's voicemail box as the ring group voicemail.
Secret	Configure the password to access the ring group extension's voicemail.
Email Address	Configure the Email address of the ring group extension's voicemail. If "Attach Recordings to E-mail" is enabled from Web GUI->**PBX**->**Voicemail**->**Voicemail Email Settings**, the voicemail can be sent to the ring group's Email address as attachment.

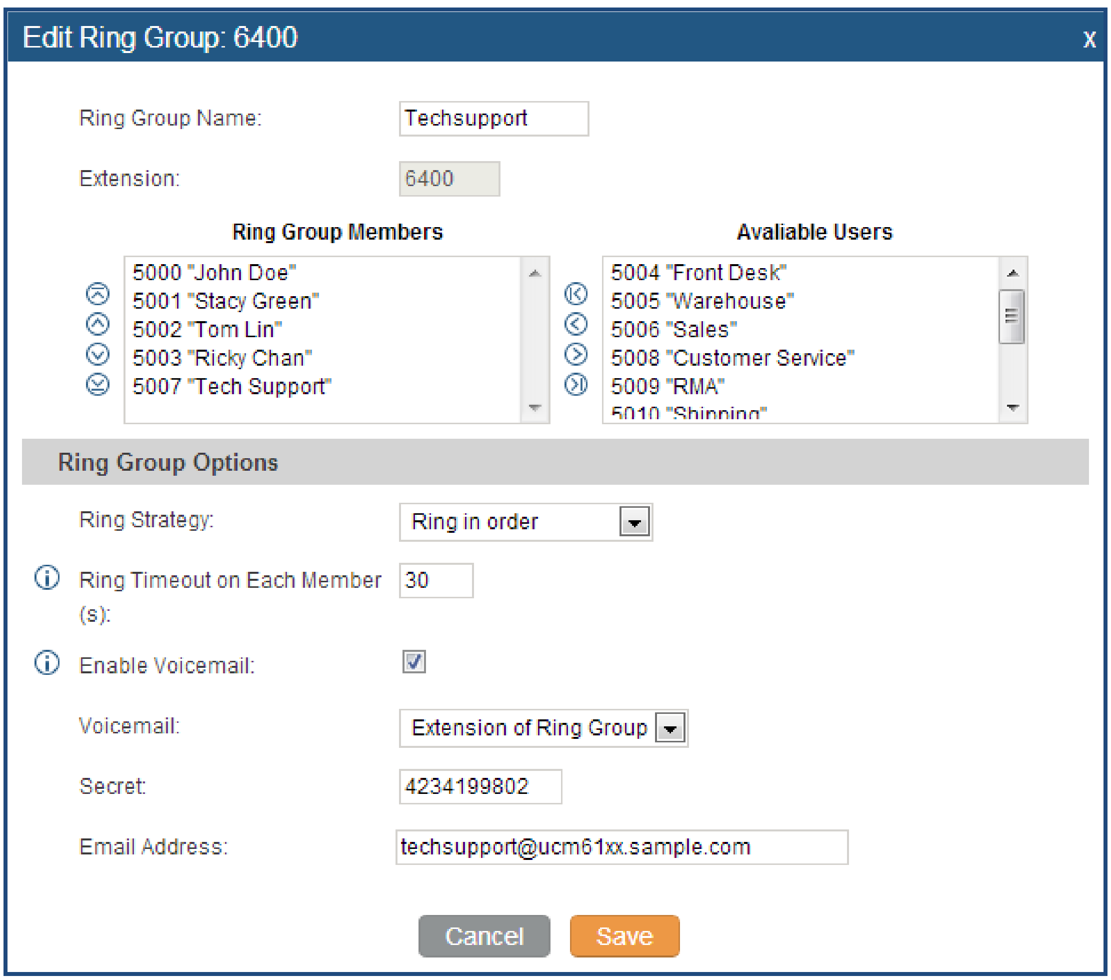


Figure 37: Ring Group Configuration

PAGING AND INTERCOM GROUP

The UCM61xx paging and intercom can be used via feature code to a single extension or a paging/intercom group. This sections describes the configuration of paging/intercom group under Web GUI->**PBX**->**Call Features**->**Paging/Intercom**.

CONFIGURE PAGING/INTERCOM GROUP

- Click on "Create New Paging/Intercom Group" to add paging/intercom group.

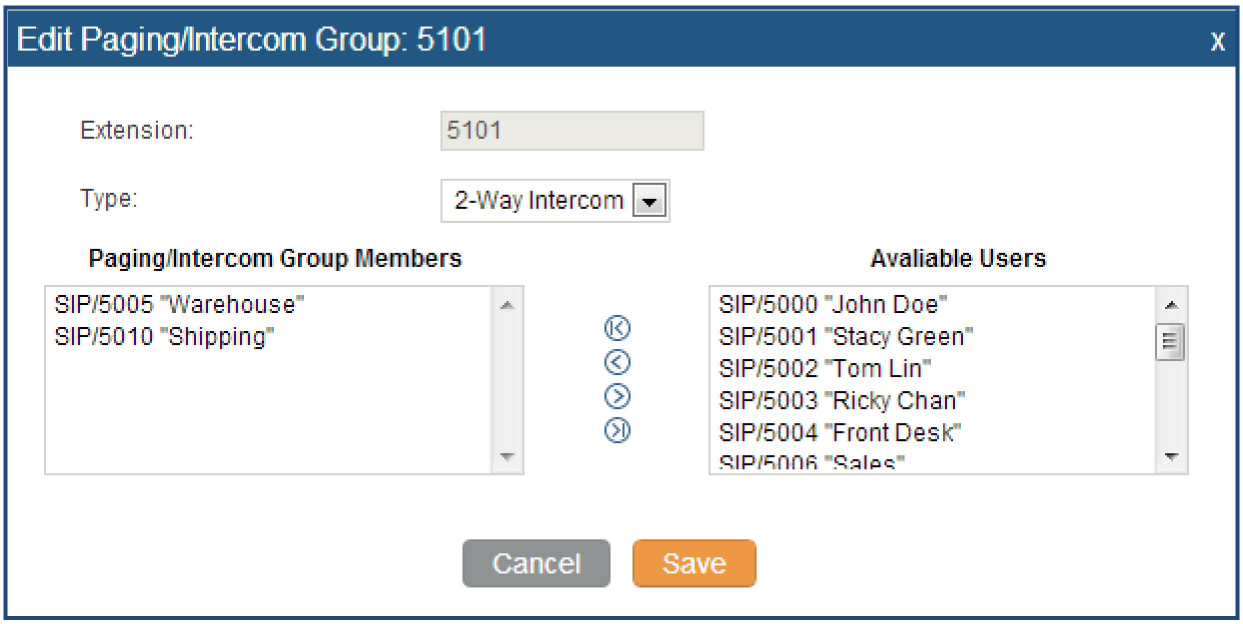


Figure 38: Page/Intercom Group

Table 33: Page/Intercom Group Configuration Parameters

Extension	Configure page/intercom group extension.
Type	Select "2-way Intercom" or "1-way Page".
Page/Intercom Group Members	Select available users from the right side to the paging/intercom group member list on the left.

- Click on ✎ to edit the page/intercom group.

- Click on 🗑 to delete the page/intercom group.

- Click on "Paging/Intercom Group Settings" to edit Alert-Info Header. This header will be included in the SIP INVITE message sent to the callee in paging/intercom call.

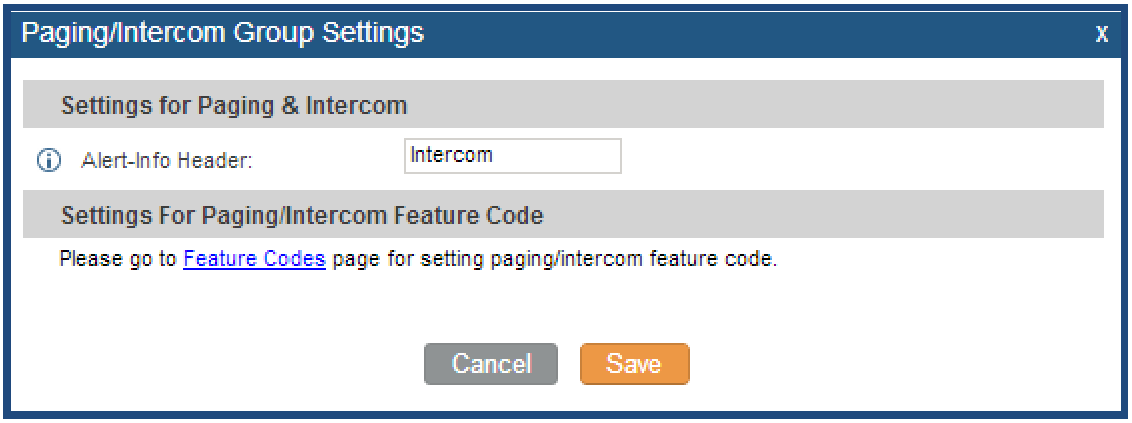


Figure 39: Page/Intercom Group Settings

- The UCM61xx has pre-configured paging/intercom feature code. To edit page/intercom feature code, click on "Feature Codes" in the "Paging/Intercom Group Settings" dialog. Or users could go to Web GUI->**PBX**->**Internal Options**->**Feature Codes** directly.

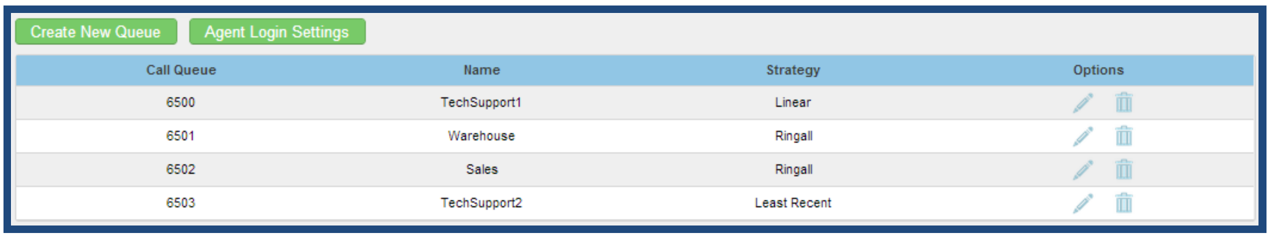
CALL QUEUE

The UCM61xx supports call queue by using static agents or dynamic agents. This sections describes the configuration of call queue under Web GUI->**PBX**->**Call Features**->**Call Queue**.

CONFIGURE CALL QUEUE

Call queue settings can be accessed via Web GUI->**PBX**->**Call Features**->**Call Queue**.

- Click on "Create New Queue" to add call queue.



Call Queue	Name	Strategy	Options
6500	TechSupport1	Linear	
6501	Warehouse	Ringall	
6502	Sales	Ringall	
6503	TechSupport2	Least Recent	

Figure 40: Call Queue

- Click on to edit the call queue. The call queue configuration parameters are listed in the table below.

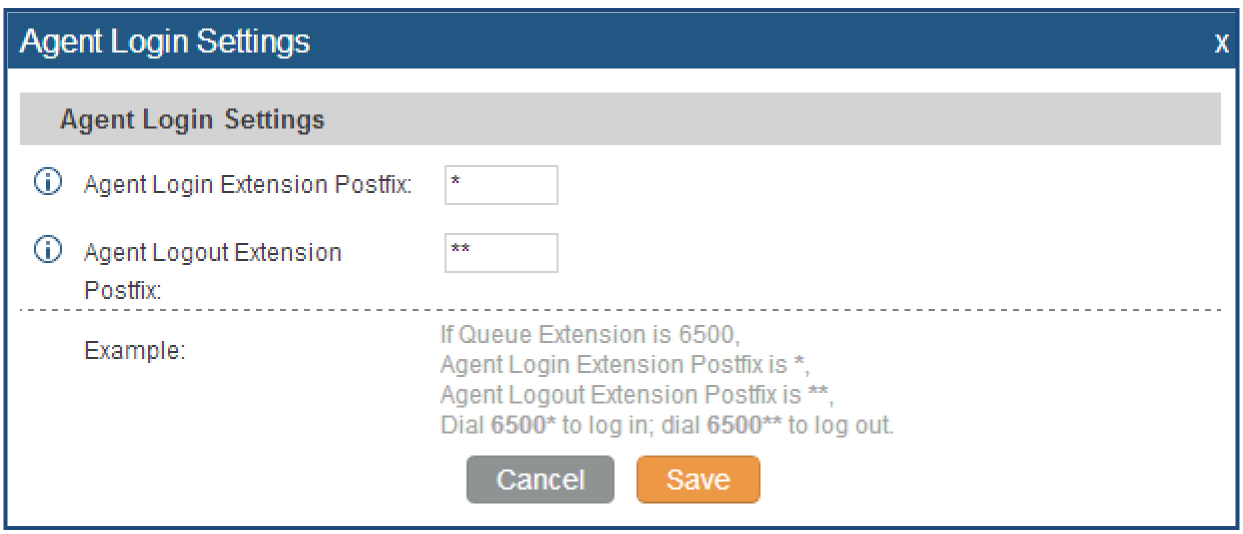
Table 34: Call Queue Configuration Parameters

Extension	Configure the call queue extension.
Name	Configure the call queue name to identify the call queue.
Strategy	Select the strategy for the call queue. • Ring All Ring all available Agents simultaneously until one answers. • Linear Ring agents in the specified order. • Least Recent Ring the agent who has been called the least recently. • Fewest Calls Ring the agent with the fewest completed calls. • Random Ring a random agent. • Round Robin Ring the agents in Round Robin scheduling with memory.

	The default setting is "Ring All".
Music On Hold	Select the Music On Hold class for the call queue. **Note:** Music On Hold classes can be managed from Web GUI->**PBX**->**Internal Options**->**Music On Hold**.
Leave When Empty	Configure whether the callers will be disconnected from the queue or not if the queue has no agent anymore. The default setting is "Strict". • Yes Callers will be disconnected from the queue if all agents are paused or invalid. • No Never disconnect the callers from the queue when the queue is empty. • Strict Callers will be disconnected from the queue if all agents are paused, invalid or unavailable.
Dial in Empty Queue	Configure whether the callers can dial into a call queue if the queue has no agent. The default setting is "No". • Yes Callers can always dial into a call queue. • No Callers cannot dial into a queue if all agents are paused or invalid. • Strict Callers cannot dial into a queue if the agents are paused, invalid or unavailable.
Dynamic Login Password	If enabled, the configured PIN number is required for dynamic agent to log in. The default setting is disabled.
Time Out	Configure the number of seconds an agent will ring before the call goes to the next agent. The default setting is 15 seconds.
Wrapup Time	Configure the number of seconds before a new call can ring the queue after the last call on the agent is completed. If set to 0, there will be no delay between calls to the queue.
Max Queue Length	Configure the maximum number of calls to be queued at once. This number does not include calls that have been connected with agents. It only includes calls not connected yet. The default setting is 0, which means unlimited. When the maximum value is reached, the caller will be treated with busy tone followed by the next calling rule after attempting to enter the queue.

Report Hold Time	If enabled, the UCM61xx will report (to the agent) the duration of time of the call before the caller is connected to the agent.
Wait Time	If enabled, users will be disconnected after the configured number of seconds. The default setting is "No". **Note:** It is recommended to configure "Wait Time" longer than the "Wrapup Time".
Static Agents	Select the available agents from the available users on the right to the static agents list on the left. Click on to arrange the order.

- Click on to delete the call queue.
- Click on "Agent Login Settings" to configure Agent Login Extension Postfix and Agent Logout Extension Postfix. Once configured, users could log in the call queue as dynamic agent.



For example, if the call queue extension is 6500, Agent Login Extension Postfix is * and Agent Logout Extension Postfix is **, users could dial 6500* to login to the call queue as dynamic agent and dial 6500** to logout from the call queue. Dynamic agent doesn't need to be listed as static agent and can log in/log out at any time.

- Call queue feature code "Agent Pause" and "Agent Unpause" can be configured under Web GUI->**PBX**->**Internal Options**->**Feature Codes**.

MUSIC ON HOLD

Music On Hold settings can be accessed via Web GUI->**PBX**->**Internal Options**->**Music On Hold**. In this page, users could configure music on hold class and upload music files. The "default" Music On Hold class already has 5 audio files defined for users to use.

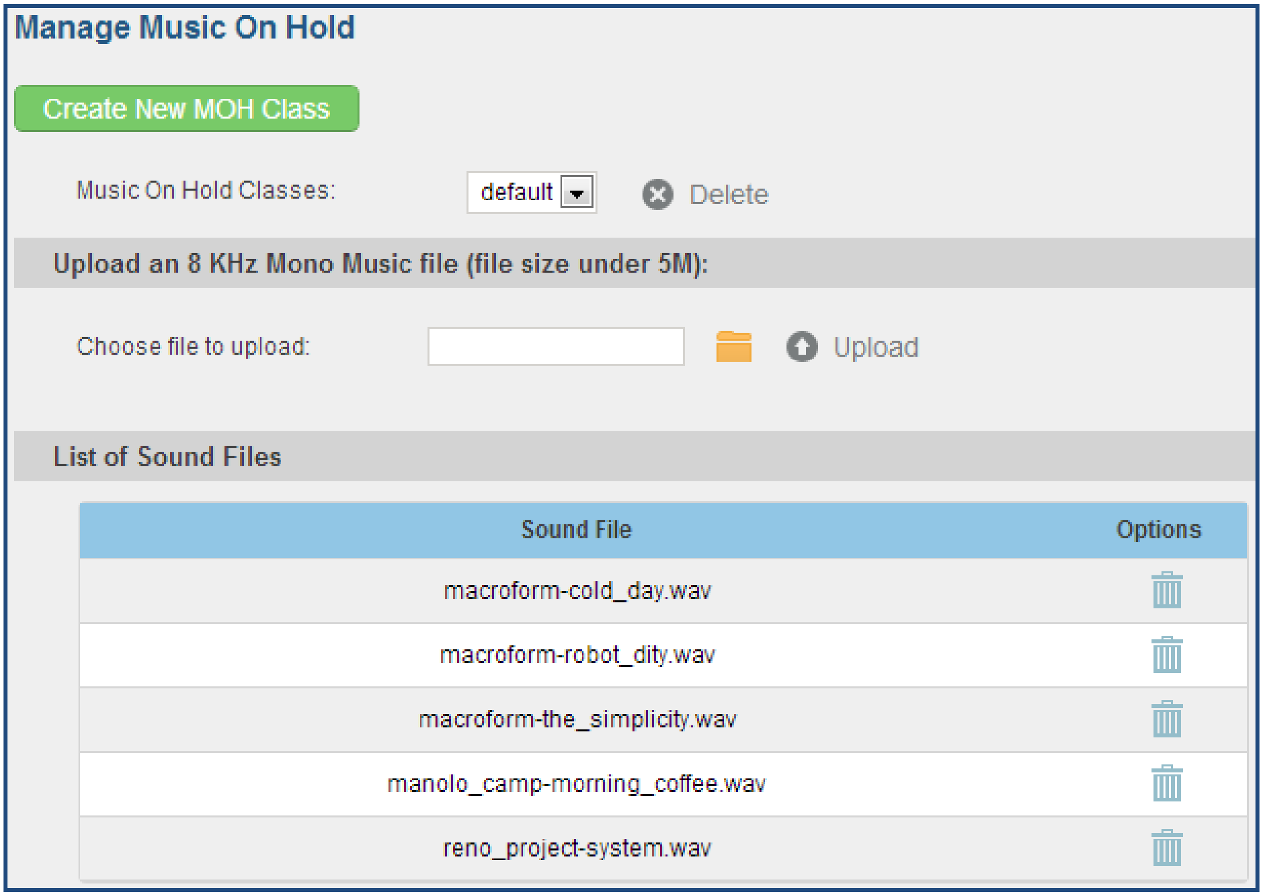


Figure 41: Music On Hold Default Class

- Click on "Create New MOH Class" to add a new Music On Hold class.
- Click on to delete the selected Music On Hold class.
- Click on to select music file from local PC and click on to start uploading. The music file uploaded has to be 8 KHz Mono format with size smaller than 5M.
- Click on to delete the sound file for the Music On Hold Class from the list of sound files.

FAX/T.38

The UCM61xx supports T.30/T.38 Fax and Fax Pass-through. After receiving the Fax, UCM61xx can convert it to PDF format and send it to the configured Email address. To do this, users could turn on "Fax Detection" for a specific VoIP trunk under UCM61xx web GUI->**PBX**->**Basic/Call Routes**->**VoIP Trunks**. Or users can set up the extension for Fax and then configure **PBX**->**Call Features**->**IVR**->**Key Pressing Events** to have the key pressing event go to the extension of the Fax.

Fax/T.38 settings can be accessed via Web GUI->**PBX**->**Internal Options**->**FAX/T.38**.

CONFIGURE FAX/T.38

- Click on "Create New Fax Extension". In the popped up window, fill the extension, name and Email address to send the received Fax to.
- Click on "Fax Settings" to configure the Fax parameters.

Table 35: FAX/T.38 Settings

Enable Error Correction Mode (ECM)	Configure to enable Error Correction Mode (ECM) for the Fax.
Maximum Transfer Rate	Configure the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14400. The default setting is 14400.
Minimum Transfer Rate	Configure the minimum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14000. The default setting is 2400.
Default Email Address	Configure the Email address to send the received Fax to if user's Email address cannot be found. **Note:** The extension's Email address or the Fax's default Email address needs to be configured in order to receive Fax from Email. If neither of them is configured, Fax will be not be received from Email.
Template Variables	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users. The template variables are:

	• ${CALLERIDNUM} : Caller ID Number • ${CALLERIDNAME} : Caller ID Name • ${RECEIVEEXTEN} : The extension to receive the Fax • ${FAXPAGES} : Number of pages in the Fax • ${VM_DATE} : The date and time when the Fax is received

- Click on to edit the Fax extension.
- Click on to delete the Fax extension.

CALL FEATURES

The UCM61xx supports call recording, transfer, call forward, call park and other call features via feature code. This section lists all the feature codes in the UCM61xx and describes how to use the call features.

FEATURE CODES

Table 36: UCM61xx Feature Codes

Feature Maps	
Blind Transfer	• Default code: #1. • Enter the code during active call. After hearing "Transfer", you will hear dial tone. Enter the number to transfer to. Then the user will be disconnected and transfer is completed. • Options **Disable** **Allow Caller**: Enable the feature code on caller side only. **Allow Callee**: Enable the feature code on callee side only. **Allow Both**: Enable the feature code on both caller and callee.
Attended Transfer	• Default code: *2. • Enter the code during active call. After hearing "Transfer", you will hear the dial tone. Enter the number to transfer to and the user will be connected to this number. Hang up the call to complete the attended transfer. • Options **Disable** **Allow Caller**: Enable the feature code on caller side only. **Allow Callee**: Enable the feature code on callee side only. **Allow Both**: Enable the feature code on both caller and callee.
Call Park	• Default code: #72. • Enter the code during active call to park the call. • Options **Disable** **Allow Caller**: Enable the feature code on caller side only. **Allow Callee**: Enable the feature code on callee side only. **Allow Both**: Enable the feature code on both caller and

	callee.
Audio Mix Record	• Default code: *3. • Enter the code followed by # or SEND to start recording the audio call and the UCM61xx will mix the streams natively on the fly as the call is in progress. • Options **Disable** **Allow Caller**: Enable the feature code on caller side only. **Allow Callee**: Enable the feature code on callee side only. **Allow Both**: Enable the feature code on both caller and callee.
DND/Call Forward	
Do Not Disturb (DND) Activate	• Default code: *77.
Do Not Disturb (DND) Deactivate	• Default code: *78.
Call Forward Busy Activate	• Default Code: *90. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Busy Deactivate	• Default Code: *91.
Call Forward No Answer Activate	• Default Code: *92. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward No Answer Deactivate	• Default Code: *93.
Call Forward Unconditional Activate	• Default Code: *72. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Unconditional Deactivate	• Default Code: *73.
Feature Misc	
Feature Code Digits Timeout	• Default Setting: 1000. • Configure the maximum interval (in milliseconds) between the digits input to activate the feature code.
Call Park	• Default Extension: 700. • During an active call, initiate blind transfer and then enter this code to park the call.
Parked Lots	• Default Extension: 701-720. • These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.

Parking Timeout (s)	• Default setting: 300. • This is the timeout allowed for a call to be parked. After the timeout, if the call is not picked up, the extension who parks the call will be called back.
Feature Codes	
Voicemail Access Code	• Default Code: *98. • Enter *98 and follow the voice prompt. Or dial *98 followed by the extension and # to access the entered extension's voicemail box.
My Voicemail	• Default Code: *97. • Press *97 to access the voicemail box.
Agent Pause	• Default Code: *83. • Pause the agent in all call queues.
Agent Unpause	• Default Code: *84. • Unpause the agent in all call queues.
Paging Prefix	• Default Code: *81. • To page an extension, enter the code followed by the extension number.
Intercom Prefix	• Default Code: *80. • To intercom an extension, enter the code followed by the extension number.
Blacklist Add	• Default Code: *40. • To add a number to blacklist for inbound route, dial *40 and follow the voice prompt to enter the number.
Blacklist Remove	• Default Code: *41. • To remove a number from current blacklist for inbound route, dial *41 and follow the voice prompt to remove the number.
Call Pickup on Ringing	• Default Code: **. • To pick up a call for extension xxxx, enter the code followed by the extension number xxxx.

CALL RECORDING

The UCM61xx allows users to record audio during the call. Please follow the instructions below to record the call.

- Make sure the feature code for "Audio Mix Record" is configured and enabled.

- After establishing the call, enter the "Audio Mix Record" feature code (by default it's *3) followed by # or SEND to start recording.
- To stop the recording, enter the "Audio Mix Record" feature code (by default it's *3) followed by # or SEND again. Or the recording will be stopped once the call hangs up.
- The recording file can be retrieved under Web GUI->**Status**->**CDR**. Click on to play the recording or click on to download the recording file.

Figure 42: Download Recording File From CDR Page

CALL PARK

The UCM61xx provides call park and call pickup features via feature code.

PARK A CALL

There are two feature codes that can be used to park the call.

- Feature Maps->Call Park (Default code #72)
  During an active call, press #72 and the call will be parked. Parking lot number (default range 701 to 720) will be announced after parking the call.

- Feature Misc->Call Park (Default code 700)
  During an active call, initiate blind transfer (default code #1) and then dial 700 to park the call. Parking lot number (default range 701 to 720) will be announced after parking the call.

RETRIEVE THE PARKED CALL

To retrieve the parked call, simply dial the parking lot number and the call will be established. If a parked call is not retrieved after the timeout, the original extension who parks the call will be called back.

INTERNAL OPTIONS

This section describes internal options that haven't been mentioned in previous sections yet. The settings in this section can be applied globally to the UCM61xx, including general configurations, jitter buffer, RTP settings, hardware config and STUN monitor. The options can be accessed via Web GUI->**PBX**->**Internal Options**.

INTERNAL OPTIONS/GENERAL

Table 37: Internal Options/General

General Preferences	
Global OutBound CID	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
Global OutBound CID Name	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
Operator Extension	Specify the operator extension, which will be dialed when users presses 0 to exit voicemail application. The operator extension can also be used in IVR option.
Ring Timeout	Configure the number of seconds to ring an extension before the call goes to the user's voicemail box. The default setting is 60.
Extension Preferences	
Enable Random Password	If enabled, random password will be generated when the extension is created. The default setting is "Yes". It is recommended to enable it for security purpose.
Disable Extension Range	If set to "Yes", users could disable the extension range pre-configured/configured on the UCM61xx. The default setting is "No". The default extension range assignment is: • User Extension: 5000-6299 • Conference Extension: 6300-6399 • IVR Extension: 7000-7100 • Ring Group Extension: 6400-6499 • Queue Extensions: 6500-6599 • VoiceMail Group Extension: 6600-6699

	Note: It is recommended to keep the system assignment to avoid inappropriate usage and unnecessary issues.

INTERNAL OPTIONS/RTP SETTINGS

Table 38: Internal Options/Jitter Buffer

SIP Jitter Buffer	
Enable Jitter Buffer	Select to enable jitter buffer on the sending side of the SIP channel. The default setting is "No".
Force Jitter Buffer	Select to force the use of jitter buffer on the receiving side of the SIP channel. The default setting is "No".
Log Frames	Select to enable jitter buffer frame logging. The default setting is "No".
Max Jitter Buffer	Configure the maximum time (in ms) to buffer for "Adaptive" jitter buffer implementation, or used as the jitter buffer size for "Fixed" jitter buffer implementation. The default setting is 200.
Resync Threshold	Configure the resync threshold for jitter buffer. When the jitter buffer notices a significant change to delay that continues over a few frames, it will resync, assuming that the change in delay is caused by a time-stamping mix-up. The threshold for noticing a change in delay is calculated as twice the measured jitter plus this resync threshold. The default setting is 1000.
Implementation	Configure the jitter buffer implementation on the sending side of a SIP channel. The default setting is "Fixed". • **Fixed** The size is always equal to the value of "Max Jitter Buffer". • **Adaptive** The size is adjusted automatically and the maximum value equals to the value of "Max Jitter Buffer".
Analog Jitter Buffer	
Enable Jitter Buffer	Select to enable jitter buffer on the receiving side of the analog channel. The default setting is "Yes".
Force Jitter Buffer	Select to force the use of jitter buffer on the receiving side of the SIP channel. The default setting is "Yes".
Log Frames	Select to enable jitter buffer frame logging. The default setting is "No".
Max Jitter Buffer	Configure the maximum time (in ms) to buffer for "Adaptive" jitter buffer

	implementation, or used as the jitter buffer size for "Fixed" jitter buffer implementation. The default setting is 200.
Resync Threshold	Configure the resync threshold for jitter buffer. When the jitter buffer notices a significant change to delay that continues over a few frames, it will resync, assuming that the change in delay is caused by a time-stamping mix-up. The threshold for noticing a change in delay is calculated as twice the measured jitter plus this resync threshold. The default setting is 1000.
Implementation	Configure the jitter buffer implementation on the receiving side of a analog channel. The default setting is "Fixed". • **Fixed** The size is always equal to the value of "Max Jitter Buffer". • **Adaptive** The size is adjusted automatically and the maximum value equals to the value of "Max Jitter Buffer".
IAX Jitter Buffer	
Enable Jitter Buffer	Select to enable jitter buffer for IAX. The default setting is "No".
Force Jitter Buffer	Select to force the use of jitter buffer on all IAX connections. The default setting is "No".
Drop Count	The drop count is the maximum number of voice packets to allow to drop (out of 100). Usually the useful value is between 3 to 10.
Max Jitter Buffer	Configure the maximum time (in ms) to buffer in the jitter. The default setting is 1000.
Max Interpolation Frames	Configure the number of interpolated frames the jitter buffer should return consecutively. The default setting is 10.
Resync Threshold	Configure the resync threshold for jitter buffer. When the jitter buffer notices a significant change to delay that continues over a few frames, it will resync, assuming that the change in delay is caused by a time-stamping mix-up. The threshold for noticing a change in delay is calculated as twice the measured jitter plus this resync threshold. If the interval is longer than the resync threshold time, resync the jitter buffer. The default setting is 1000.
Max Excess Buffer	Configure the maximum amount of excess jitter buffer (in milliseconds) to pad to the jitter buffer before the jitter buffer is slowly shrunk to eliminate latency.
Min Excess Buffer	Configure the minimum amount of excess jitter buffer (in milliseconds) to pad to the jitter buffer before the jitter buffer to slowly raised to eliminate latency.

Jitter Shrink Rate	Configure the rate at which the jitter buffers are increased or decreased.

INTERNAL OPTIONS/RTP SETTINGS

Table 39: Internal Options/RTP Settings

RTP Start	Configure the RTP port starting number. The default setting is 10000.
RTP End	Configure the RTP port ending address. The default setting is 20000.
Strict RTP	Configure to enable or disable strict RTP protection. If enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
RTP Checksums	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".

INTERNAL OPTIONS/HARDWARE CONFIG

The analog hardware (FXS port and FXO port) on the UCM61xx will be listed in this page. Click on to edit signaling preference for FXS port or configure ACIM settings for FXO port.

Select "Loop Start" or "Kewl Start" for each FXS port. And then click on "Update" to save the change.

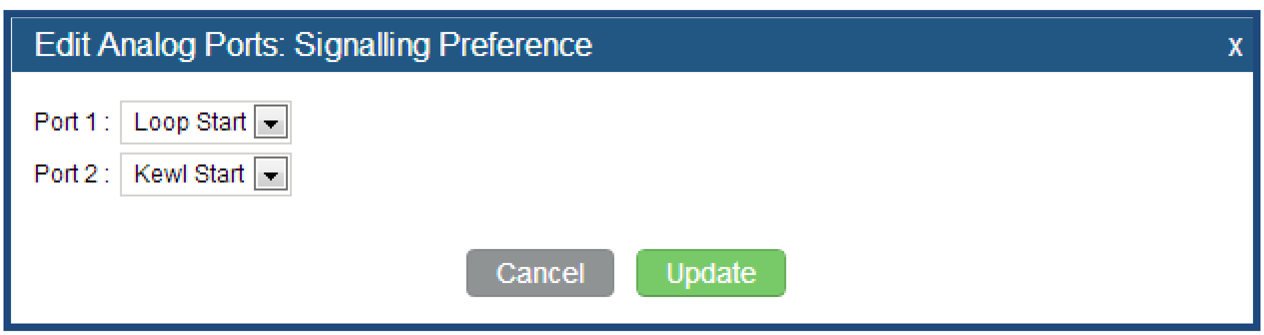


Figure 43: FXS Ports Signaling Preference

For FXO port, users could manually enter the ACIM settings by selecting the value from dropdown list for each port. Or users could click on "Detect" for the UCM61xx to automatically detect the ACIM value. The detecting value will be automatically filled into the settings.

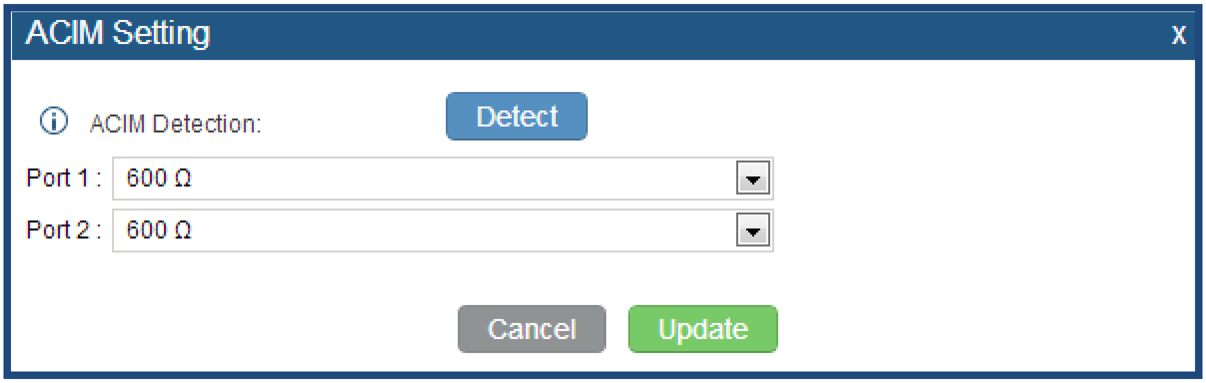


Figure 44: FXO Ports ACIM Settings

Table 40: Internal Options/Hardware Config

Tone Region	Select country to set the default tones for dial tone, busy tone, ring tone and etc to be sent from the FXS port. The default setting is "United States of America (USA)".
Advanced Settings	
FXO Opermode	Select country to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
FXS Opermode	Select country to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "United States of America (USA)".
TISS Override	Select the impedance value for Two-Wire Impedance Synthesis (TISS) override.
PCMA Override	Select the codec to be used for analog lines. North American users should choose PCMU. All other countries, unless already known, should be assumed to be PCMA. The default setting is PCMU. **Note:** This option requires system reboot to take effect.
Boost Ringer	Configure whether normal ringing voltage (40V) or maximum ringing voltage (89V) for analog phones attached to the FXS port is required. The default setting is "Normal".
Fast Ringer	Configure to increase the ringing speed to 25HZ. This option can be

	used with "Low Power" option. The default setting is "Normal".
Low Power	Configure the peak voltage up to 50V during "Fast Ringer" operation. This option is used with "Fast Ringer". The default setting is "Normal".
Ring Detect	If set to "Full Wave", false ring detection will be prevented for lines where Caller ID is sent before the first ring and proceeded by a polarity reversal, as in UK. The default setting is "Standard".
MWI Mode	Configure the type of Message Waiting Indicator detection on trunk (FXO) interfaces. The default setting is "None". • None: No detection • FSK: Frequency Shift Key detection • NEON: Neon MWI detection

INTERNAL OPTIONS/STUN MONITOR

Table 41: Internal Options/STUN Monitor

STUN Server	Configures the IP address or URL of the STUN server to query. If not specified, STUN is disabled. The default setting is stun.ipvideotalk.com. Valid format: [(hostname | IP-address) [':' port] The default port number is 3478 if not specified.
STUN Refresh	Configure the number of seconds between STUN Refreshes. The default setting is 30 seconds.

IAX SETTINGS

The UCM61xx IAX global settings can be accessed via Web GUI->**PBX**->**IAX Settings**.

IAX SETTINGS/GENERAL

Table 42: IAX Settings/General

Bind Port	Configure the port number that the IAX2 will be allowed to listen to. The default setting is 4569.
Bind Address	Configure the address that the IAX2 will be forced to bind to. The default setting is 0.0.0.0, which means all addresses.
IAX1 Compatibility	Select to configure IAX1 compatibility. The default setting is "No".
No Checksums	If selected, UDP checksums will be disabled and no checksums will be calculated/checked on systems supporting this features. The default setting is "No".
Delay Reject	If enabled, the IAX2 will delay the rejection of calls to avoid DOS. The default setting is "No".
ADSI	Select to enable ADSI phone compatibility. The default setting is "No".
Music On Hold Interpret	Specify which Music On Hold class this channel would like to listen to when being put on hold. This music class is only effective if this channel has no music class configured and the bridged channel putting the call on hold has no "Music On Hold Suggest" setting.
Music On Hold Suggest	Specify which Music On Hold class to suggest to the bridged channel when putting the call on hold.
Bandwidth	Configure the bandwidth for IAX settings. The default setting is "Low".

IAX SETTINGS/CODECS

The following codes are supported in UCM61xx for IAX. Select the codecs from the right side list to the left side. Click on to arrange the order.

- PCMU
- PCMA
- GSM
- ILBC
- G.722
- G.726

- ADPCM
- LPC10
- G.729
- G.723
- H.263
- H.263p
- H.264

IAX SETTINGS/REGISTRATION

Table 43: IAX Settings/Registration

Min Reg Expire	Configure the minimum period (in seconds) of registration. The default setting is 60.
Max Reg Expire	Configure the maximum period (in seconds) of registration. The default setting is 3600.
IAX Thread Count	Configure the number of IAX helper threads. The default setting is 10.
IAX Max Thread Count	Configure the maximum number of IAX threads allowed. The default setting is 100.
Auto Kill	If set to "yes", the connection will be terminated if ACK for the NEW message is not received within 2000ms. Users could also specify number (in milliseconds) in addition to "yes" and "no". The default setting is "yes".
Authentication Debugging	If enabled, authentication traffic in debugging will not show. The default setting is "No".
Codec Priority	Configure codec negotiation priority. The default setting is "Reqonly". • Caller Consider the callers preferred order ahead of the host's. • Host Consider the host's preferred order ahead of the caller's. • Disabled Disable the consideration of codec preference all together. • Reqonly This is almost the same as "Disabled", except when the requested format is not available. The call will only be accepted if the requested format is available.
Type of Service	Configure ToS bit for preferred IP routing.
Trunk Frequency	Configure the frequency of trunk frames (in milliseconds). The default

	setting is 20.
Trunk Time Stamps	If enabled, time stamps will be attached to trunk frames. The default setting is "No".

IAX SETTINGS/STATIC DEFENSE

Table 44: IAX Settings/Static Defense

Call Token Optional	Enter a single IP address or a range of IP addresses for which call token validation is not required. For example: 11.11.11.11 11.11.11.11/22.22.22.22.
Max Call Numbers	Configure the maximum number of calls allowed for a single IP address.
Max Nonvalidated Call Numbers	Configure the maximum number of unvalidated calls for all IP addresses.
Call Number Limits	Configure to limit the number of calls for a give IP address of IP range.
IP or IP Range	Enter the IP address or a range of IP addresses to be considered for call number limits. For example: 11.11.11.11 11.11.11.11/22.22.22.22.

SIP SETTINGS

The UCM61xx SIP global settings can be accessed via Web GUI->**PBX**->**SIP Settings**.

SIP SETTINGS/GENERAL

Table 45: SIP Settings/General

Realm For Digest Authentication	Configure the host name or domain name for the UCM61xx. Realms MUST be globally unique according to RFC3261. The default setting is Grandstream.
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IP Address	Configure the IP address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
Allow Guest Calls	If enabled, the UCM61xx allows unauthorized INVITE coming into the PBX and the call can be made. The default setting is "No".
Overlap Dialing Support	Select to enable overlap dialing support. The default setting is "No".
Allow Transfer	If set to "No", all transfers initiated by the endpoint in the UCM61xx will be disabled (unless enabled in peers or users). The default setting is "Yes".
Enable DNS SRV Lookups on Outbound Calls	Select to enables DNS SRV lookups on outbound calls from the UCM61xx. The default setting is "Yes".
MWI From	When sending MWI NOTIFY requests, this value will be used in the "From:" header as the "name" field. If no "From User" is configured, the "user" field of the URI in the "From:" header will be filled with this value.
Domain	Configure the domain for the UCM61xx. Incoming INVITE and REFER messages can be matched against a list of "allowed" domains, each of which can direct the call to a specific context if desired. By default, all domains are accepted and sent to the default context or the context associated with the user/peer placing the call. Register to non-local domains will be automatically denied if a domain list is configured. Up to 10 domains can be added.
From Domain	Configure the domain in the "From:" header of the SIP message. It may be required by some providers for authentication.
Auto Domain	If enabled, the UCM61xx will add local host name and local IP to domain list. The default setting is "No".
Allow External Domains	If enabled, requests for external domains that are not served by the

	UCM61xx will be allowed. The default setting is "Yes".

SIP SETTINGS/CODECS

The following codecs are supported in UCM61xx for SIP. Select the codecs from the right side list to the left side. Click on to arrange the order as appeared in the SDP of the SIP message.

- PCMU
- PCMA
- GSM
- ILBC
- G.722
- G.726
- ADPCM
- LPC10
- G.729
- G.723
- H.263
- H.263p
- H.264

SIP SETTINGS/MISC

Table 46: SIP Settings/Misc

Register Timeout	Configure the register retry timeout (in seconds). The default setting is 20.
Register Attempts	Configure the number of registration attempts before the UCM61xx gives up. The default setting is 0, which means the UCM61xx will keep trying until the server side accepts the registration request.
Video Max Bit Rate (kb/s)	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
Support for SIP Video	Select to enable video support in SIP calls. The default setting is "Yes".
Generate Manager Events	If enabled, the UCM61xx will generate manager events when SIP UA performs events (e.g. Hold). The default setting is "No".
Reject Non-Matching Invites	If enabled, when rejecting an incoming INVITE or REGISTER request, the UCM61xx will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP

	usernames. The default setting is "No".
Non-Standard G.726 Support	If enabled, when the peer negotiates G726-32 audio, the UCM61xx will use AAL2 packing order instead of RFC3551 packing order (AAL2-G726-32). The default setting is "No".

SIP SETTINGS/SESSION TIMER

Table 47: SIP Settings/Session Timer

Session Timers	Select the session timer mode. The default setting is "Accept". The options are: • Originate Always request and run session timer. • Accept Run session timer only when requested by other UA. • Refuse Do not run session timer.
Session Expires	Configure the maximum session refresh interval (in seconds). The default setting is 1800.
Min SE	Configure the minimum session refresh interval (in seconds). The default setting is 90.
Session Refresher	Select the session refresher to be UAC or UAS. The default setting is UAC.

SIP SETTINGS/TCP and TLS

Table 48: SIP Settings/TCP and TLS

TCP Enable	Configure to allow incoming TCP connections with the UCM61xx. The default setting is "No".
TCP Bind Address	Configure the IP address for TCP server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5060 will be used.
TLS Enable	Configure to allow incoming TLS connections with the UCM61xx. The default setting is "No".
TLS Bind Address	Configure the IP address for TLS server to bind to. 0.0.0.0 means binding to all interfaces. The port number is optional. If not specified, 5061 will be used.

	Note: The IP address must match the common name (hostname) in the certificate. Please do not bind a TLS socket to multiple IP addresses. For details on how to construct a certificate for SIP, please refer to the following document: http://tools.ietf.org/html/draft-ietf-sip-domain-certs
TLS Client Protocol	Select the TLS protocol for outbound client connections. The default setting is TLSv1.
TLS Do Not Verify	If enabled, the TLS server's certificate won't be verified when acting as a client. The default setting is "Yes".
TLS Self-Signed CA	This is the CA certificate if the TLS server being connected to requires self-signed certificate, including server's public key. This file will be renames as "TLS.ca" automatically. **Note:** The size of the uploaded ca file must be under 2MB.
TLS Cert	This is the Certificate file (*.pem format only) used for TLS connections. It contains private key for client and signed certificate for the server. This file will be renamed as "TLS.pem" automatically. **Note:** The size of the uploaded certificate file must be under 2MB.
TLS CA Cert	This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers. **Note:** The size of the uploaded CA certificate file must be under 2MB.
TLS CA List	Display a list of files under the CA Cert directory.

SIP SETTINGS/TCP and TLS

Table 49: SIP Settings/NAT

External IP Address	Configure a static address and port (optional) that will be used in outbound SIP messages if the UCM61xx is behind NAT. If it's a hostname, it will only be looked up once.
External Host	Specify an external host name, which is similar to External Address except the host name will be looked up periodically based on the

	"External Refresh" interval.
External Refresh	Configure the refresh interval for the external host (if used) The default setting is 10.
External TCP Port	Configure the externally mapped TCP port when the UCM61xx is behind a static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when UCM61xx is behind a static NAT or PAT.
Local Network Address	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/16
NAT Mode	This is a global NAT setting that will affects all peers and users. The default setting is "Force rport". • No: Use rport if the remote side requires it. • Force rport: Force rport to always be on. • Yes: Force rport to be always on and perform comedia RTP handling. • Comedia: Use rport if the remote side requires it and performs comedia RTP handling. **Note:** "comedia RTP handling" refers to the technique of sending RTP to the port where the other endpoint's RTP packets come from. This can also be rephrased as "connection-oriented media".
Allow RTP Reinvite	If enabled, the UCM61xx will try to redirect the RTP media stream (audio) to go directly from the caller to the callee. The default setting is "No NAT". • Yes • No NAT: Allow media path redirection (Reinvite) but only when the peer is not be behind NAT. The RTP core can detect if the peer is behind NAT or not based on the IP address where the media comes from. • Update: Use UPDATE for media path redirection, instead of INVITE. **Note:** Some devices do not support this (especially if one of them is behind NAT).

SIP SETTINGS/TOS

Table 50: SIP Settings/TOS

ToS For SIP	Configure the Type of Service for SIP packets. The default setting is None.
ToS For RTP Audio	Configure the Type of Service for RTP audio packets. The default setting is None.
ToS For RTP Video	Configure the Type of Service for RTP video packets. The default setting is None.
Default Incoming/Outgoing Registration Time	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.
Max Registration/Subscription Time	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the UCM61xx. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the UCM61xx. The default setting is 60.
Music On Hold Interpret	Configure the Music On Hold class for the channel when being put on hold. This is used when the Music On Hold class is not set on the channel and the peer channel placing the call on hold doesn't have "Music On Hold Suggest".
Music On Hold Suggest	Configure the Music On Hold class to suggest to the peer channel when placing the peer on hold.
Enable Relaxed DTMF	Select to enable relaxed DTMF handling. The default setting is "No".
DTMF Mode	Select DTMF mode to send DTMF. The default setting is RFC2833. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, "RFC2833" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC2833".
RTP Timeout	During an active call, if there is no RTP activity within the timeout (in seconds), the call will be terminated. The default setting is no timeout. **Note:** This setting doesn't apply to calls on hold.
RTP Hold Timeout	When the call is on hold, if there is no RTP activity within the timeout (in seconds), the call will be terminated. This value of RTP Hold Timeout should be larger than RTP Timeout. The default setting is no timeout.
Trust Remote Party ID	Configure whether the Remote-Party-ID should be trusted. The default setting is "No".

Send Remote Party ID	Configure whether the Remote-Party-ID should be sent or not. The default setting is "No".
Generate In-Band Ringing	Configure whether the UCM61xx should generate inband ringing or not. The default setting is "Never". • Yes: The UCM61xx will send 180 Ringing followed by 183 Session Progress and in-band audio. • No: The UCM61xx will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing. • Never: Whenever ringing occurs, the UCM61xx will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly.
Server User Agent	Configure the user agent string for the UCM61xx.
Allow Non-local Redirect	If enabled, 302 or REDIRECT is allowed to non-local SIP address. The default setting is "No".
Add "user=phone" to URI	If enabled, "user=phone" will be added to URI that contains a valid phone number. The default setting is "No".
Send Compact SIP Headers	If enabled, compact SIP headers will be sent. The default setting is "No".
MWI Checking Interval	Configure the default interval (in seconds) for checking MWI status of peer's voicemail. The default setting is 10.
Min SIP T1 Timeout	Configure the minimum roundtrip time (in milliseconds) for the SIP messages sent to the monitored hosts. The default setting is 100.

SIP SETTINGS/DEBUG

Table 51: SIP Settings/Debug

Enable SIP Debugging	Select to enable SIP debugging. The default setting is "No".
Record SIP History	Select to enable recording SIP history. The default setting is "No".
Dump SIP History	Select to enable dump SIP history at the end of SIP dialogue. The default setting is "No".
Subscribe Context	Configure a specific context for SUBSCRIBE requests. This setting is useful to limit subscriptions to local extensions. The default setting is "from-internal".
Allow Subscribe	Configure to allow subscriptions. The default setting is "Yes".
Notify on Ringing	Configure to send out NOTIFY on ringing state. The default setting is "Yes".

STATUS AND REPORTING

PBX STATUS

The UCM61xx monitors the status for Trunks, Extensions, Queues, Conference Rooms, Interfaces and Parking lot. It presents administrators the real time status in different sections under web GUI->**Status**->**PBX Status**.

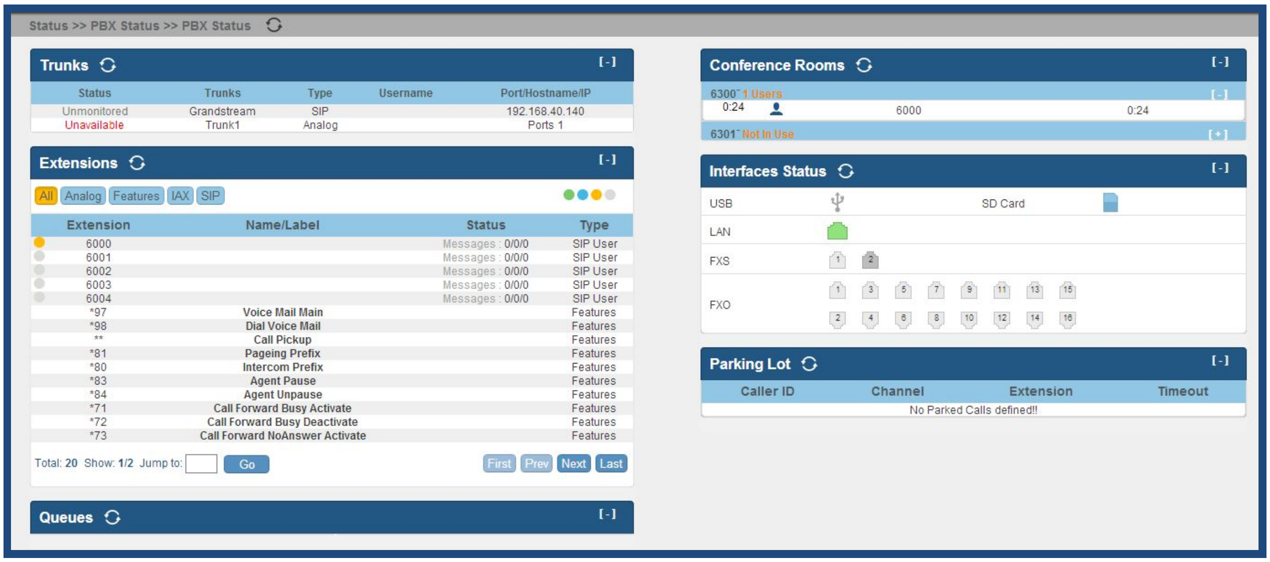


Figure 45: Status->PBX Status

TRUNKS

Users could see all the configured trunk status in this section.

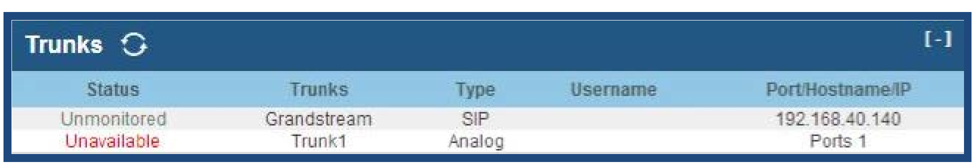


Figure 46: Trunk Status

Table 52: Trunk Status

Status	Display trunk status. • Analog trunk status: **Available** **Busy** **Unavailable** **Unknown Error** • SIP Peer trunk status: **Unreachable**: The hostname cannot be reached. **Unmonitored**: QUALIFY feature is not turned on to be monitored. **Reachable**: The hostname can be reached. • SIP Register trunk status: **Registered** **Unrecognized Trunk**
Trunks	Display trunk name
Type	Display trunk Type: • Analog • SIP • IAX
Username	Display username for this trunk.
Port/Hostname/IP	Display Port for analog trunk, or Hostname/IP for VoIP (SIP/IAX) trunk.

Other operations are also available in trunk status section:

- Click on "Trunks", the web page will redirect to trunk configuration page which can also be accessed via web GUI->**PBX**->**Basic/Call Routes**->**Analog Trunks**.
- Click on to refresh the trunk status.
- Click on [+] to expand the status detail table.
- Click on [-] to hide the status detail table.

EXTENSIONS

Users could see all the configured extension status in this section.

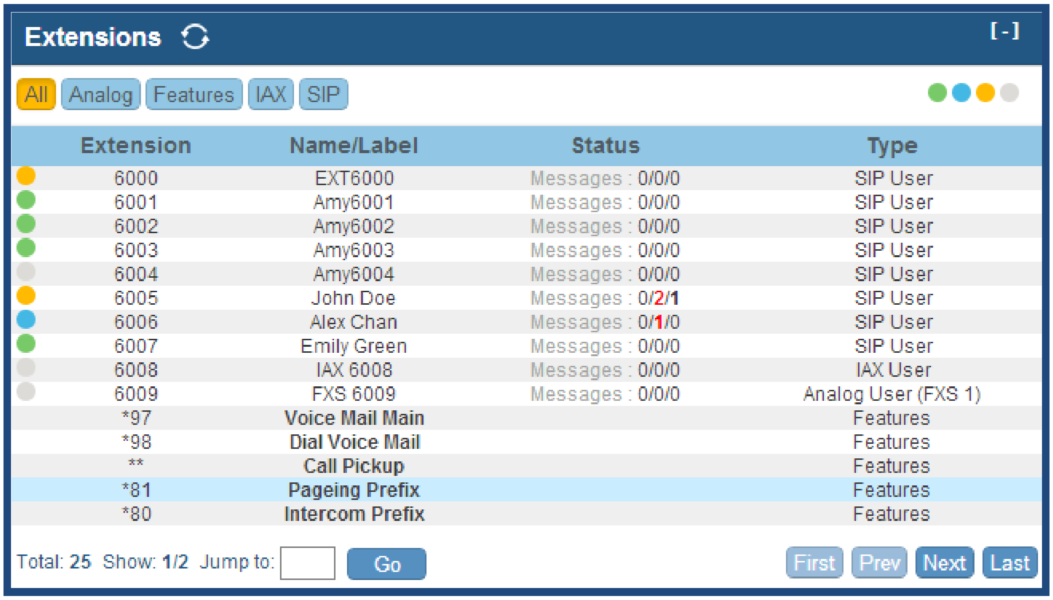


Figure 47: Extension Status

Table 53: Extension Status

Extension	Display extension number (including feature code). The color indicator has the following definitions. • Green: Free • Blue: Ringing • Yellow: In Use • Grey: Unavailable
Name/Label	Display name (callerID name) or label for the extension.
Status	Display message status for the extension. Example: 2/4/1 Description: There are 2 urgent messages, 4 messages in total and 1 message that has been already read.
Type	Displays extension type. • SIP User • IAX User • Analog User • Features

Other operations are also available in extension status section:

- Click on "Extensions", the web page will redirect to extension configuration page which can also be accessed via web GUI->**PBX**->**Basic/Call Routes**->**Extensions**.

- Click on to refresh the extension status.
- Click on one of the tabs All Analog Features IAX SIP to display the corresponding extensions accordingly.
- Click on [+] to expand the status detail table.
- Click on [-] to hide the status detail table.

QUEUES

Users could see all the configured call queue status in this section. The following figure shows the call queue 6500 being in used.

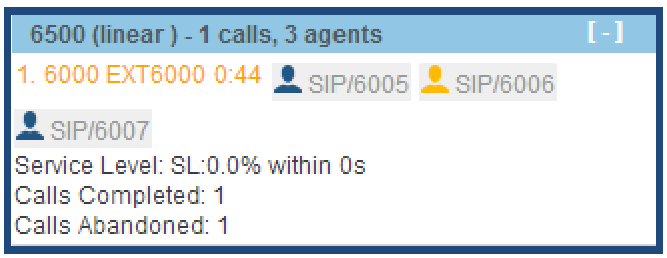


Figure 48: Queue Status

The current call status (caller ID, duration), agent status, service level, calls summary (completed/abandoned) are shown for the call queue. The agent status is defined as below.

Table 54: Agent Status

	The agent is available/idle.
	The agent is talking/busy.
	The agent is ringing.
	The agent has been logged out.

On the UCM61xx, **Service Level** is defined as the percentage of high-quality calls over all calls in the call queue, where high-quality call means calls answered within 10 seconds.

Other operations are also available in queue status section:

- Click on "Queues", the web page will redirect to call queue configuration page which can also be accessed via web GUI->**PBX**->**Call Features**->**Call Queue**.
- Click on to refresh the call queue status.

- Click on [+] to expand the call queue detail.
- Click on [-] to hide the call queue detail.

CONFERENCE ROOMS

Users could see all the conference room status in this section. It shows all the configured conference rooms, current users, call duration for each user and conference call.

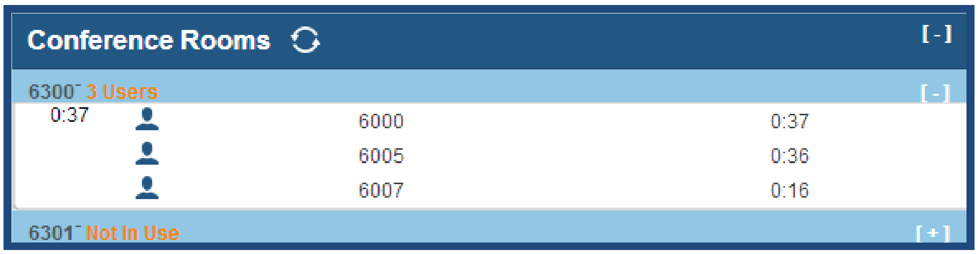


Figure 49: Conference Room Status

Other operations are also available in conference room status section:

- Click on "Conference Rooms", the web page will redirect to conference room configuration page which can also be accessed via web GUI->**PBX**->**Call Features**->**Conference**.
- Click on to refresh the conference room status.
- Click on [+] to expand the conference room details.
- Click on [-] to hide the conference room details.

INTERFACES STATUS

This section displays interface/port connection status on the UCM61xx. The following example shows the interface status for UCM6116 with USB, SD card, LAN port and FXS1 connected.

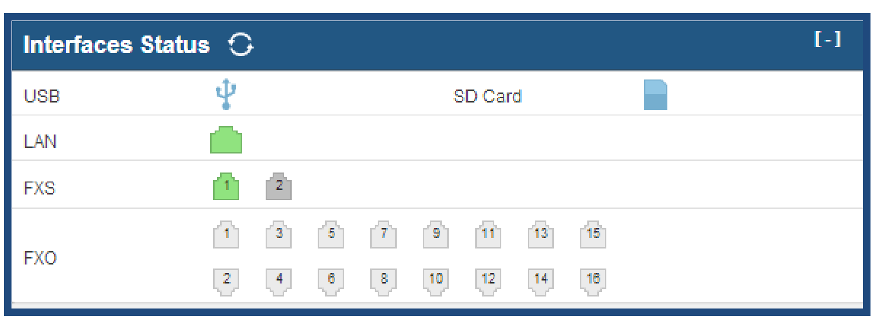


Figure 50: UCM6116 Interfaces Status

Table 55: Interface Status Indicators

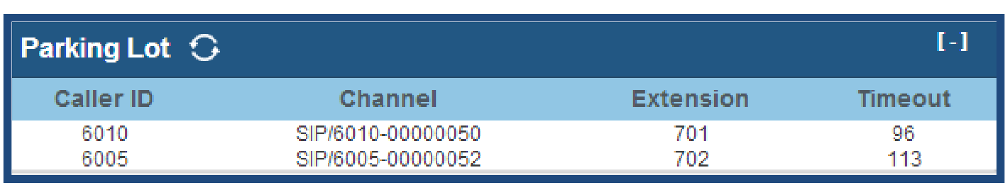
	USB connected.
	USB disconnected.
	SD Card connected.
	SD Card disconnected.
	LAN/WAN connected.
	LAN/WAN not configured.
	LAN/WAN disconnected.
	FXS/FXO connected.
	FXS/FXO waiting.
	FXS/FXO busy.
	FXS/FXO not configured.
	FXS/FXO disconnected.

Other operations are also available in interface status section:

- Click on "Interfaces Status", the web page will redirect to hardware configuration page which can also be accessed via web GUI->**PBX**->**Internal Options**->**Hardware Config**.
- Click on to refresh the interface status.
- Click on [+] to expand the interface details.
- Click on [-] to hide the interface details.

PARKING LOT

The UCM61xx supports call park using feature code. When there is call being parked, this section will display the parking lot status.



Parking Lot [-]

Caller ID	Channel	Extension	Timeout
6010	SIP/6010-00000050	701	96
6005	SIP/6005-00000052	702	113

Figure 51: Parking Lot Status

Table 56: Parking Lot Status

Caller ID	Display the caller ID who parks the call.
Channel	Display channel for the call park.
Extension	Display the parking lot number where the call is parked/retrieved.
Timeout	Display timeout (in seconds) for the parked call. The status page will dynamically update this timer from 120 seconds (default) to 0. When the timer reaches 0, the caller who parks the call will be called back.

Other operations are also available in parking lot status section:

- Click on "Parking Lot", the web page will redirect to feature codes page which can also be accessed via web GUI->**PBX**->**Internal Options**->**Feature Codes**.
- Click on to refresh the parking lot status.
- Click on [+] to expand the parking lot details.
- Click on [-] to hide the parking details.

SYSTEM STATUS

The UCM61xx system status can be accessed via Web GUI->**Status**->**System Status**, which displays the following system information.

- **General**
- **Network**
- **Storage Usage**
- **Resource Usage**

GENERAL

Under Web GUI->**Status**->**System Status**->**General**, users could check the hardware and software information for the UCM61xx. Please see details in the following table.

Table 57: System Status->General

Status ->System Status -> General	
Model	Product model.

Part Number	Product part number.
System Time	Current system time.
Up Time	System up time since the last reboot.
Idle Time	System idle time since the last reboot.
Boot	Boot version.
Core	Core version.
Base	Base version.
Program	Program version. This is the main software release version.
Recovery	Recovery version.

NETWORK

Under Web GUI->**Status**->**System Status**->**Network**, users could check the network information for the UCM61xx. Please see details in the following table.

Table 58: System Status->Network

Status -> System Status -> Network	
MAC Address	Global unique ID of device, in HEX format. The MAC address can be found on the label coming with original box and on the label located on the bottom of the device.
IP Address	IP address.
Gateway	Default gateway address.
Subnet Mask	Subnet mask address.
DNS	DNS Server address.

STORAGE USAGE

Users could access the storage usage information from Web GUI->**Status**->**System Status**->**Storage Usage**. It shows the available and used space for the following partitions.

- Configuration partition
  This partition contains PBX system configuration files and service configuration files.
- Data partition
  Voicemail, recording files, IVR file, music on hold files and etc.
- USB disk
  USB disk will display if connected.
- SD Card

SD Card will display if connected.

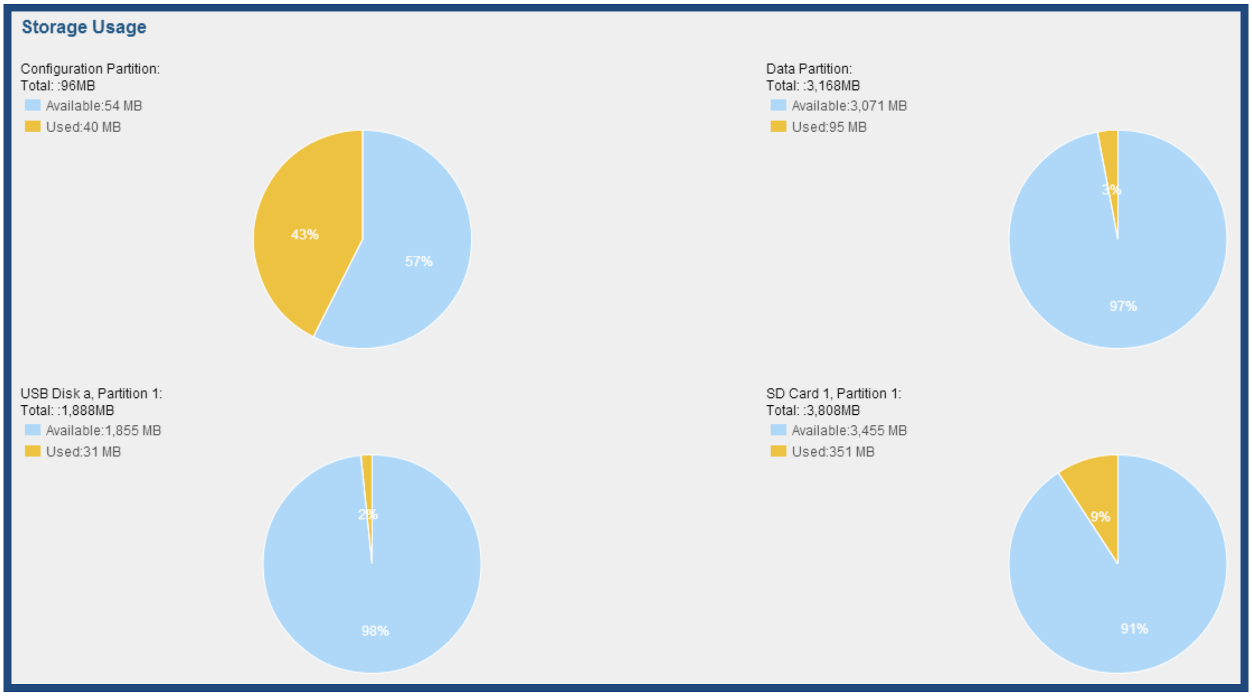


Figure 52: System Status->Storage Usage

RESOURCE USAGE

When configuring and managing the UCM61xx, users could access resource usage information to estimate the current usage and allocate the resources accordingly. Under Web GUI->**Status**->**System Status**->**Resource Usage**, the current CPU usage and Memory usage are shown in the pie chart.

Figure 53: System Status->Resource Usage

CDR (CALL DETAIL REPORT)

A Call Detail Record (CDR) is a data record produced by telephone exchange activities or other telecommunications equipment documenting the details of a phone call that passed through the PBX. The CDR is composed of the following data fields on the UCM61xx.

- **Start Time**. Format: 2013-03-27 16:47:03.
- **Call From**. Format: "John Doe"<6012>.
- **Call To**. Format: 6005.
- **Call Time**. Format: 0:00:10.
- **Talk Time**. Format: 0:00:10
- **Status**. Format: NO ANSWER, BUSY, ANSWERED, or FAILED.
- **Option**. Voice record playing/downloading/deleting.

Users could filter the call report by specifying the date range and criteria, depending on how the users would like to include the logs to the report. Then click on "View Report" button to display the generated report.

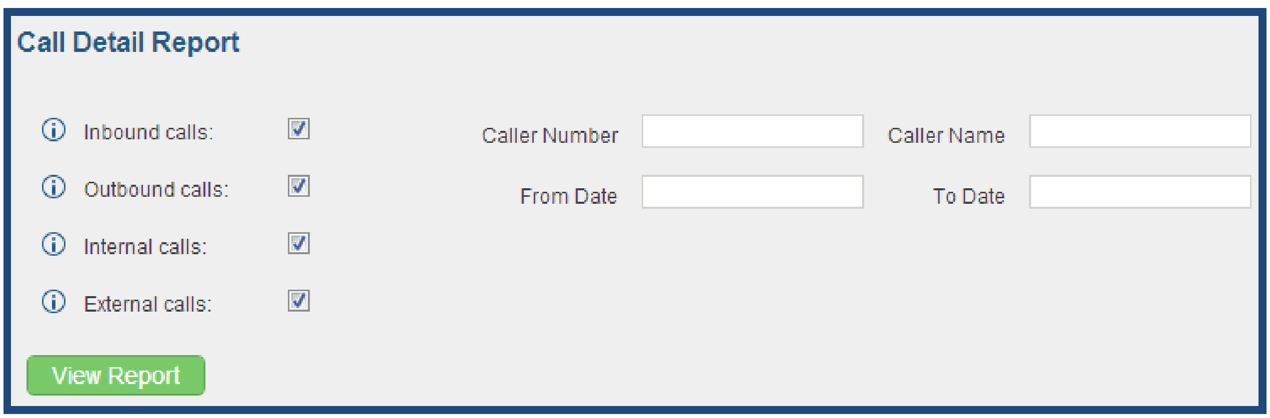


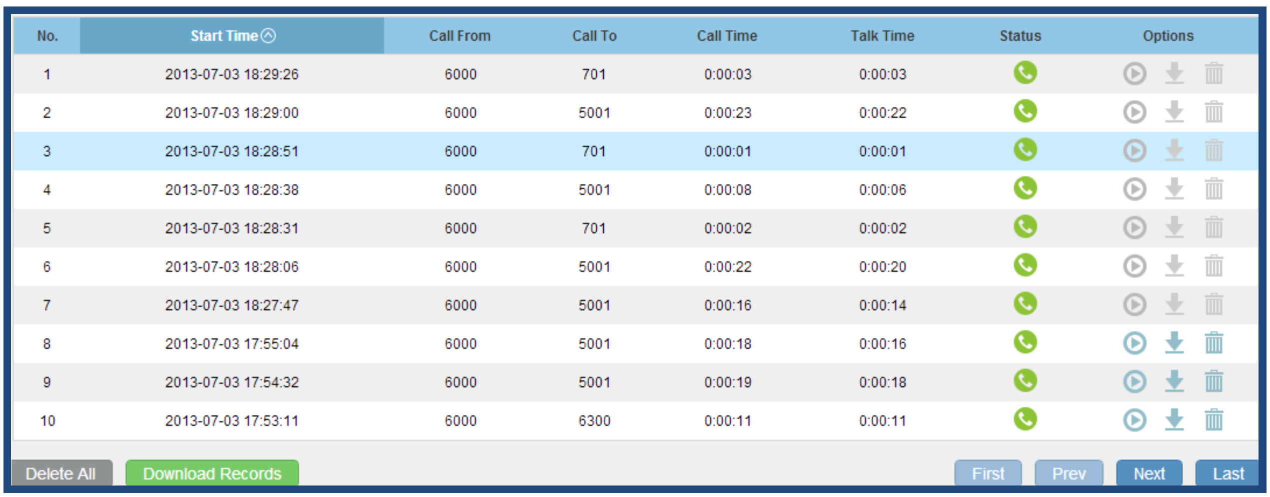
Figure 54: CDR Filter

Table 59: CDR Filter Criteria

Inbound calls	Inbound calls are calls originated from a non-internal source (like a VoIP trunk) and sent to an internal extension.
Outbound calls	Outbound calls are calls sent to a non-internal source (like a VoIP trunk) from an internal extension.
Internal calls	Internal calls are calls from one internal extension to another extension, which are not sent over a trunk.

External calls	External calls are calls sent from one trunk to another trunk, which are not sent to any internal extension.
Caller Number	Enter the caller number to be filtered in the CDR report.
Caller Name	Enter the caller name to be filtered in the CDR report.
From Date	Specify "From" date and time to be filtered for the CDR report. Click on the field and the calendar will show for users to select the exact date and time.
To Date	Specify "To" date and time to be filtered for the CDR report. Click on the field and the calendar will show for users to select the exact date and time.

The call report will display as the following figure shows.



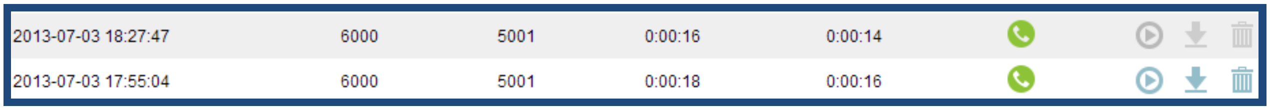
No.	Start Time	Call From	Call To	Call Time	Talk Time	Status	Options
1	2013-07-03 18:29:26	6000	701	0:00:03	0:00:03		
2	2013-07-03 18:29:00	6000	5001	0:00:23	0:00:22		
3	2013-07-03 18:28:51	6000	701	0:00:01	0:00:01		
4	2013-07-03 18:28:38	6000	5001	0:00:08	0:00:06		
5	2013-07-03 18:28:31	6000	701	0:00:02	0:00:02		
6	2013-07-03 18:28:06	6000	5001	0:00:22	0:00:20		
7	2013-07-03 18:27:47	6000	5001	0:00:16	0:00:14		
8	2013-07-03 17:55:04	6000	5001	0:00:18	0:00:16		
9	2013-07-03 17:54:32	6000	5001	0:00:19	0:00:18		
10	2013-07-03 17:53:11	6000	6300	0:00:11	0:00:11		

Figure 55: Call Report

Users could perform the following operations on the call report.

- **Sort**
  Click on the header of the column to sort by this category. For example, clicking on "Start Time" will sort the report according to start time. Clicking on "Start Time" again will reverse the order.
- **Download Records**
  On the bottom of the page, click on "Download Records" button to export the report in .csv format.
- **Delete All**
  On the bottom of the page, click on "Delete All" button to remove all the call report information.
- **Play/Download/Delete Recording File (per entry)**
  If the entry has audio recording file for the call, the three icons on the most right column will be activated for users to select. In the following picture, the second entry has audio recording file for the call.

Click on to play the recording file; click on to download the recording file in .wav format; click on to delete the recording file (the call record entry will not be deleted).



2013-07-03 18:27:47	6000	5001	0:00:16	0:00:14		
2013-07-03 17:55:04	6000	5001	0:00:18	0:00:16		

Figure 56: Call Report Entry With Audio Recording File

CDR Statistics is an additional feature on the UCM61xx which provides users a visual overview of the call report across the time frame. Users can filter with different criteria to generate the statistics chart.

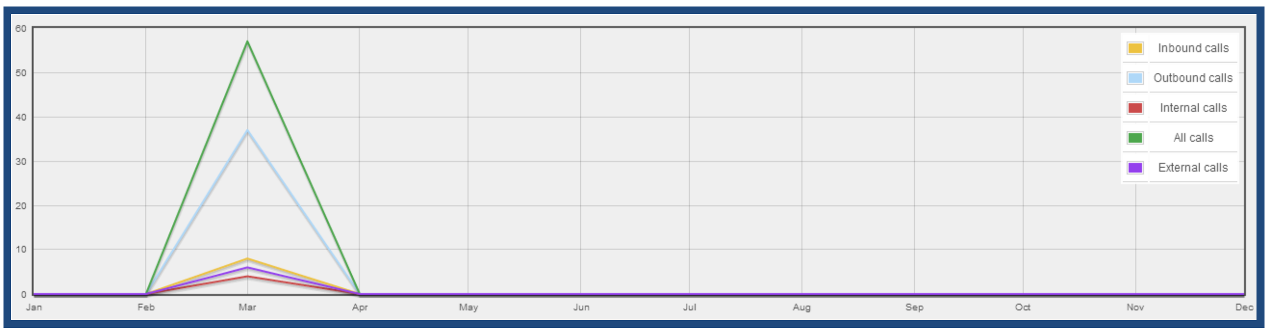


Figure 57: CDR Statistics

Table 60: CDR Statistics Filter Criteria

Trunk Type	Select one of the following trunk type. • All • SIP Calls • PSTN Calls
Call Type	Select one or more in the following checkboxes. • Inbound calls • Outbound calls • Internal calls • External calls • All calls
Time Range	• By month (of the selected year). • By week (of the selected year). • By day (of the specified month for the year). • By hour (of the specified date). • By range. For example, 2013-01 To 2013-03.

UPGRADING AND MAINTENANCE

UPGRADING

The UCM61xx can be upgraded to a new firmware version remotely or locally. This section describes how to upgrade your UCM61xx via network or local upload.

UPGRADING VIA NETWORK

The UCM61xx can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP, HTTP or HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:
firmware.grandstream.com

The upgrading configuration can be accessed via **Web GUI**->**Maintenance**->**Upgrade**.

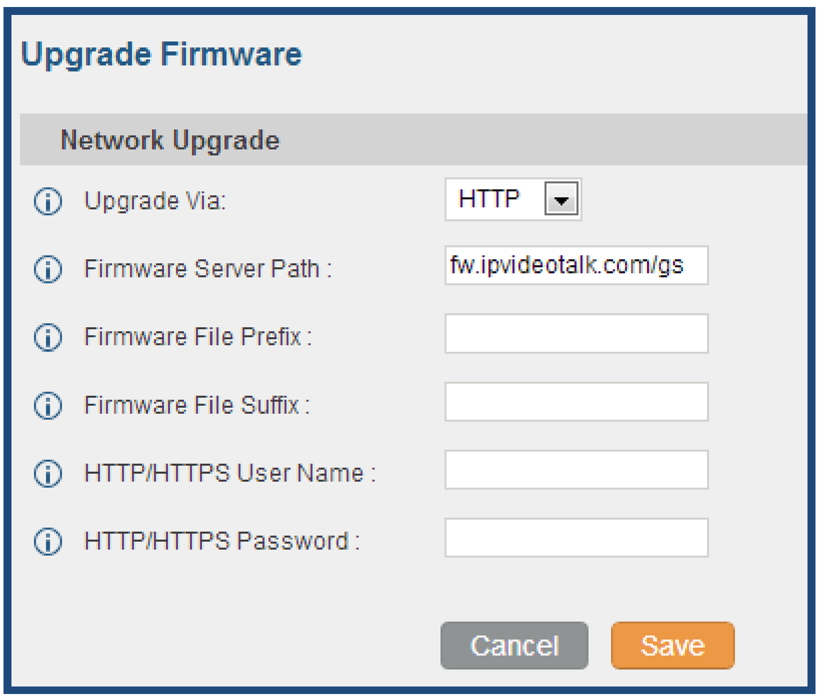


Figure 58: Network Upgrade

Table 61: Network Upgrade Configuration

Upgrade Via	Allow users to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
Firmware Server Path	Define the server path for the firmware server.
Firmware File Prefix	If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the UCM61xx.
Firmware File Suffix	If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the UCM61xx.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.

Please follow the steps below to upgrade the firmware remotely.

- Enter the firmware server path under **Web GUI**->**Maintenance**->**Upgrade**.
- Click on "Save". Then reboot the device to start the upgrading process.
- Please be patient during the upgrading process. Once done, a reboot message will be displayed in the LCD.
- Manually reboot the UCM61xx when it's appropriate to avoid immediate service interruption. After it boots up, log in the web GUI to check the firmware version.

UPGRADING VIA LOCAL UPLOAD

If there is no HTTP/TFTP server, users could also upload the firmware to the UCM61xx directly via Web GUI. Please follow the steps below to upload firmware locally.

- Download the latest UCM61xx firmware file from the following link and save it in your PC.
  http://www.grandstream.com/support/firmware
- Log in the Web GUI as administrator in the PC.
- Go to Web GUI->**Maintenance**->**Upgrade**, upload the firmware file by clicking on  and select the firmware file from your PC. The default firmware file name is ucm6100fw.bin

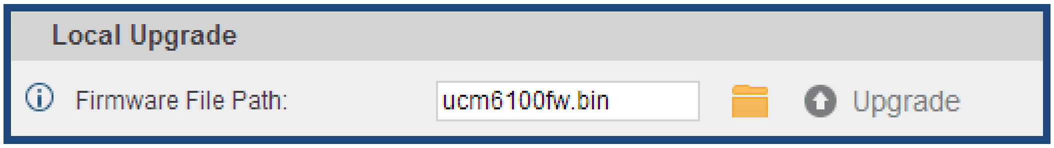


Figure 59: Local Upgrade

- Click on  to start upgrading.

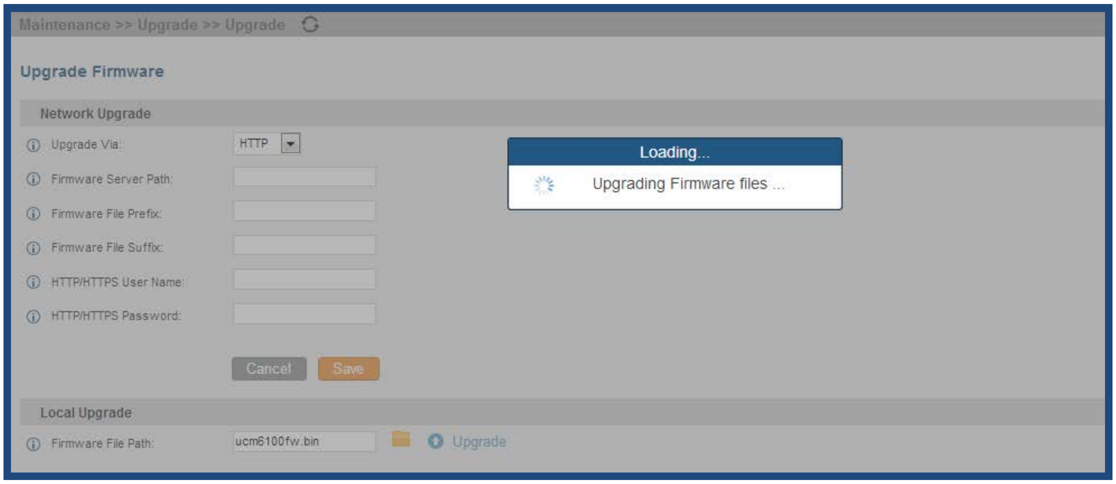


Figure 60: Upgrading Firmware Files

- Wait until the upgrading process is successful and a window will be popped up in the Web GUI.

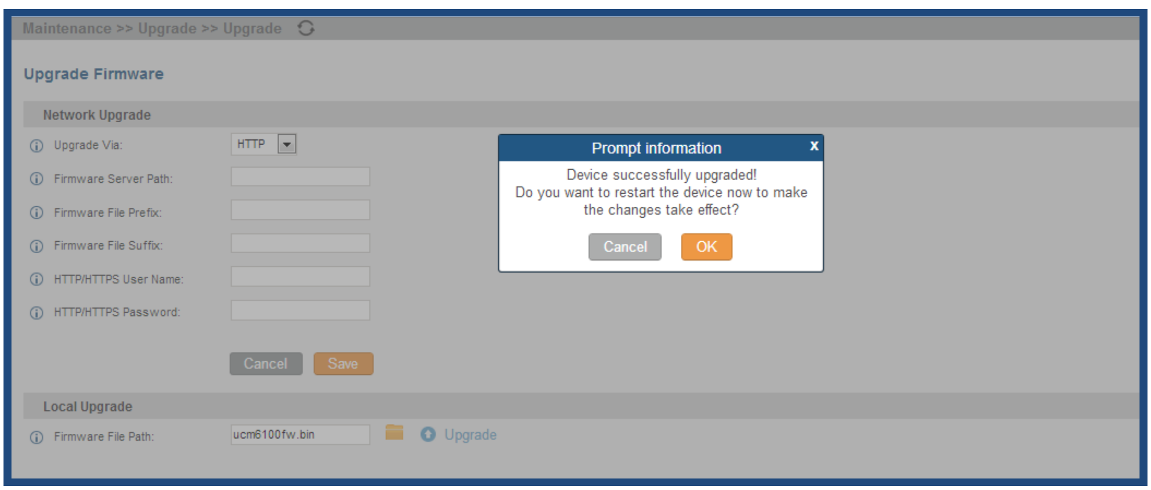


Figure 61: Reboot UCM61xx

- Click on "OK" to reboot the UCM61xx and check the firmware version after it boots up.

⚠ **Note:**

Please do not interrupt or power cycle the UCM61xx during upgrading process.

NO LOCAL FIRMWARE SERVERS

For users that would like to use remote upgrading without a local TFTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their devices via this server. Please refer to the webpage:
http://www.grandstream.com/support/firmware.

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free windows version TFTP server is available for download from :
http://www.solarwinds.com/products/freetools/free_tftp_server.aspx
http://tftpd32.jounin.net

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server;
2. Connect the PC running the TFTP server and the UCM61xx to the same LAN segment;
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade;
4. Start the TFTP server and configure the TFTP server in the UCM61xx web configuration interface;
5. Configure the Firmware Server Path to the IP address of the PC;
6. Update the changes and reboot the UCM61xx.

End users can also choose to download a free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

BACKUP

The UCM61xx configuration can be backed up locally or via network. The backup file will be used to restore the configuration on UCM61xx when necessary.

LOCAL BACKUP

Users could backup the UCM61xx configurations for restore purpose under Web GUI->**Maintenance**->**Backup**->**Local Backup**. Before creating new backup file, select the backup option first.

- If the Config-File is selected only, the backup file will be saved in the flash of the UCM61xx.
- If Voice-File, Voicemail-File, Voice-Records or CDR is selected, external storage devices (USB Flash drive or SD Card) will be required because the backup file might be too large.

Click on "Create New Backup" button to start backup. Once the backup is done, the list of the backups will be displayed with date and time in the web page. Users can download, restore, or delete it from the UCM61xx internal storage or the external device.

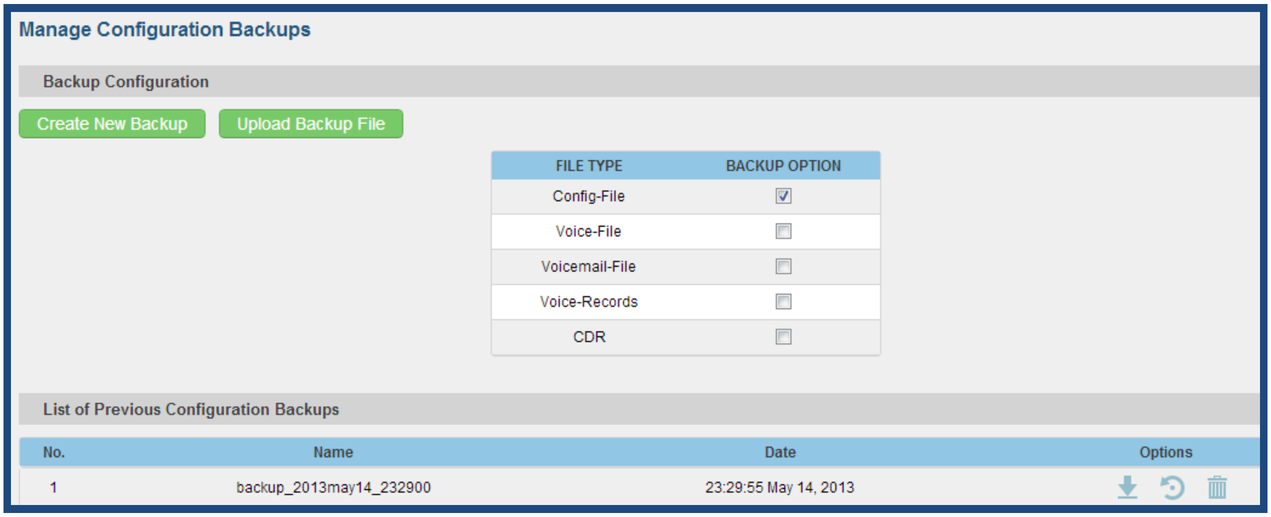


Figure 62: Local Backup

NETWORK BACKUP

Besides local backup, users could backup the voice records/voice mails/CDR/FAX in a daily basis to a remote server via SFTP protocol automatically under Web GUI->**Maintenance**->**Backup**->**Network Backup**.

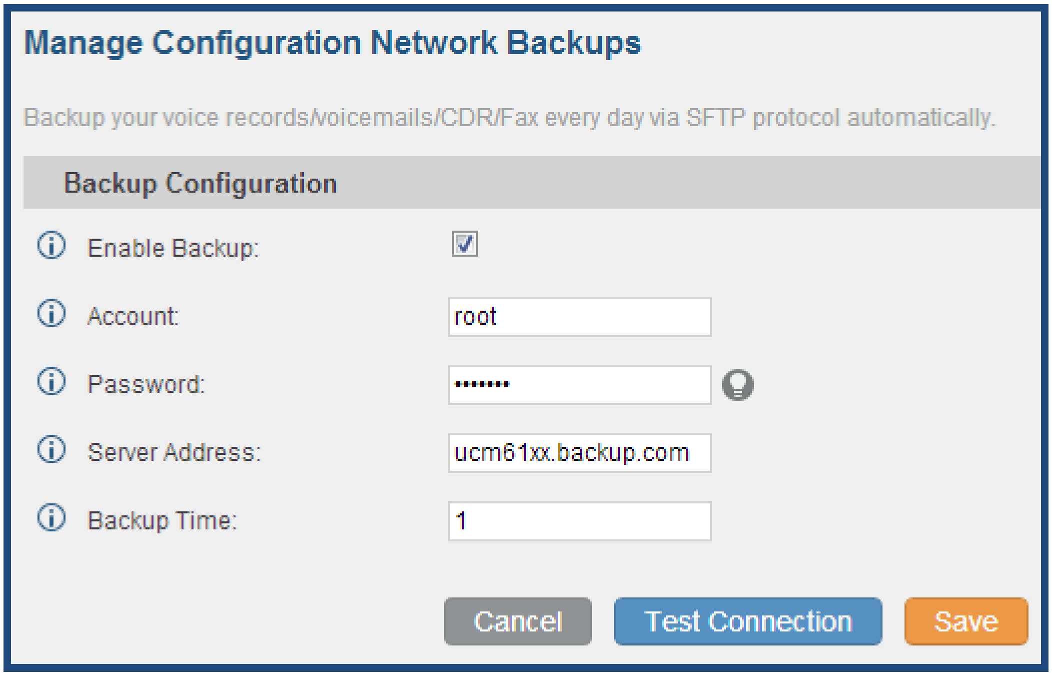


Figure 63: Network Backup

Table 62: Network Backup Configuration

Enable Backup	Enable the auto backup function. The default setting is "No".
Account	Enter the Account name on the SFTP backup server.
Password	Enter the Password associate with the Account on the SFTP backup server.
Server Address	Enter the SFTP server address.
Backup Time	Enter 0-23 to specify the backup hour of the day.

Before saving the configuration, users could click on "Test Connection". The UCM61xx will then try connecting the server to make sure the server is up and accessible for the UCM61xx.

Save the changes and all the backup logs will be listed on the web page.

RESTORE CONFIGURATION FROM BACKUP FILE

To restore the configuration on the UCM61xx from a backup file, users could go to Web GUI->**Maintenance**->**Backup**->**Local Backup**.

- A list of previous configuration backups is displayed on the web page. Users could click on  of the desired backup file and it will be restored to the UCM61xx.

- If users have other backup files on PC to restore on the UCM61xx, click on "Upload Backup File" first and select it from local PC to upload on the UCM61xx. Once the uploading is done, this backup file will be displayed in the list of previous configuration backups for restore purpose. Click on  to restore from the backup file.

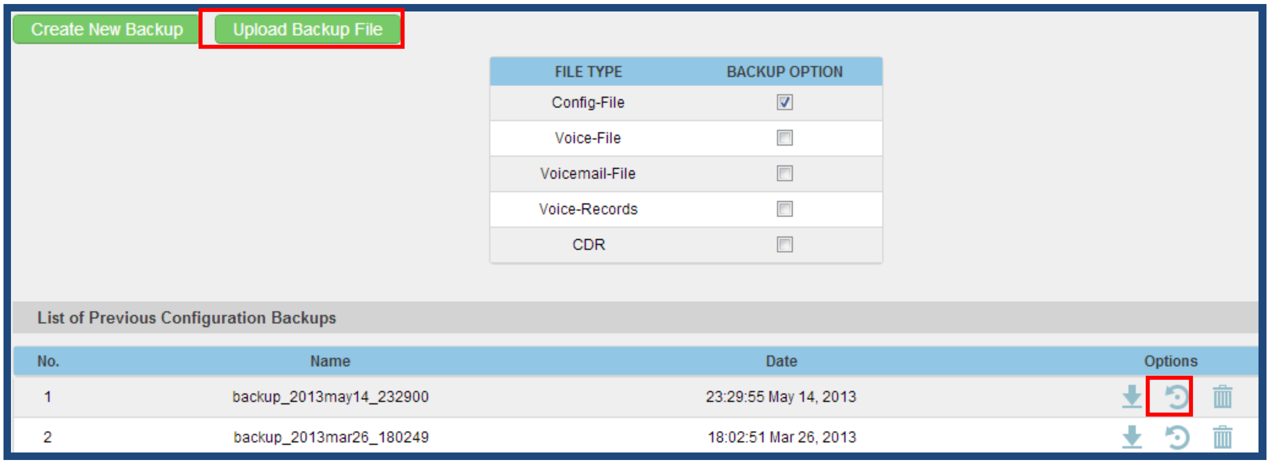


Figure 64: Restore UCM61xx From Backup File

⚠ **Note:**

- The uploaded backup file must be a tar file with no special characters like *,!,#,@,&,$,%,^,(,),/,\,space in the file name.
- The uploaded back file size must be under 10MB.

CLEANER

Users could configure to clean the Call Detail Report/Voice Records/Voice Mails/FAX automatically under Web GUI->**Maintenance**->**Cleaner**.

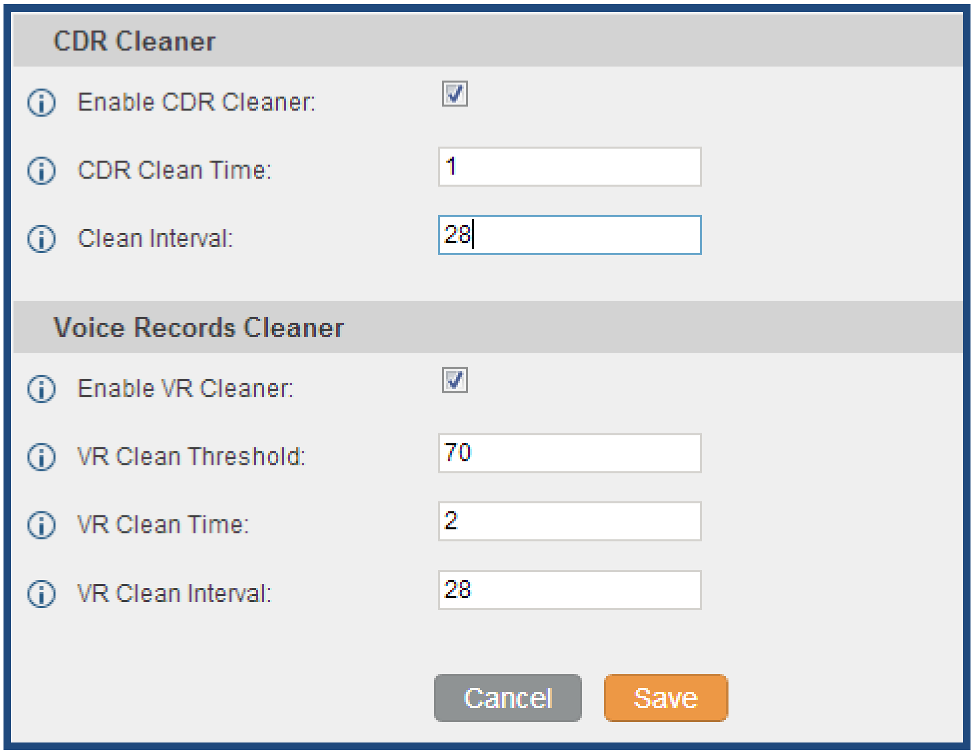


Figure 65: Cleaner

Table 63: Cleaner Configuration

Enable CDR Cleaner	Enable the CDR Cleaner function.
CDR Clean Time	Enter 0-23 to specify the hour of the day to clean up CDR.
Clean Interval	Enter 1-30 to specify the day of the month to clean up CDR.
Enable VR Cleaner	Enter the Voice Records Cleaner function.
VR Clean Threshold	Specify the Voice Records threshold from 0 to 99 by using local storage status in percentage.
VR Clean Time	Enter 0-23 to specify the hour of the day to clean up Voice Records.
Clean Interval	Enter 1-30 to specify the day of the month to clean up Voice Records.

All the cleaner logs will be listed on the bottom of the page.

RESET AND REBOOT

Users could perform reset and reboot under Web GUI->**Maintenance**->**Reset and Reboot**.
To factory reset the device, select the mode type first. There are three different types for reset.

- User Data: All the data including voicemail, recordings, IVR Prompt, Music on Hold, CDR and backup files will be cleared.
- All: All the configurations and data will be reset to factory default.

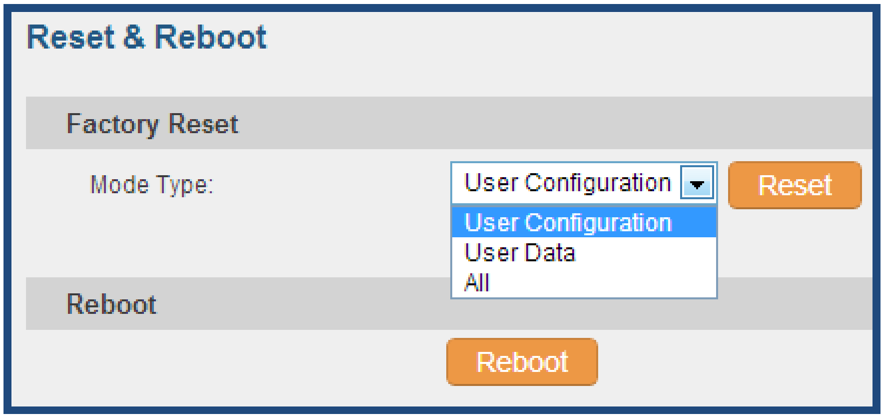


Figure 66: Reset and Reboot

SYSLOG

On the UCM61xx, users could dump the syslog information to a remote server under Web GUI->**Maintenance**->**Syslog**. Enter the syslog server hostname or IP address and select the module/level for the syslog information.

The default syslog level for all modules is "error", which is recommended in your UCM61xx settings because it can be helpful to locate the issues when errors happen.

Some typical modules for UCM61xx functions are as follows and users can turn on "notic" and "verb" levels besides "error" level.

pbx: This module is related to general PBX functions.
chan_sip: This module is related to SIP calls.
chan_dahdi: This module is related to analog calls (FXO/FXS).
app_meetme: This module is related to conference bridge.

TROUBLESHOOTING

On the UCM61xx, users could capture traces, ping remote host and traceroute remote host for troubleshooting purpose under Web GUI->**Maintenance**->**Troubleshooting**.

ETHERNET CAPTURE

The captured trace can be downloaded for analysis. Also the instructions or result will be displayed in the web GUI output result.

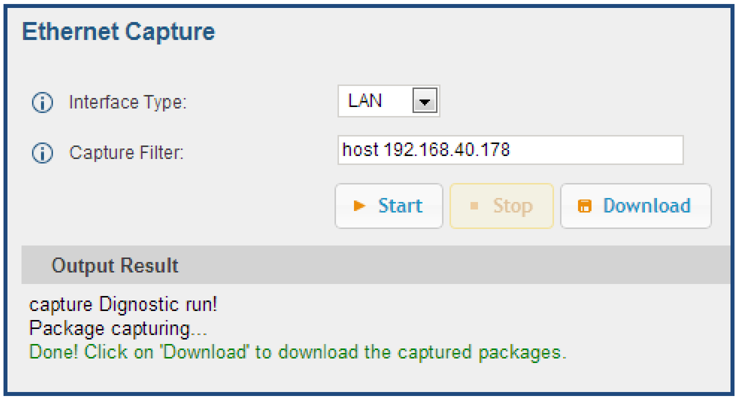


Figure 67: Ethernet Capture

The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting capturing the trace.

PING

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

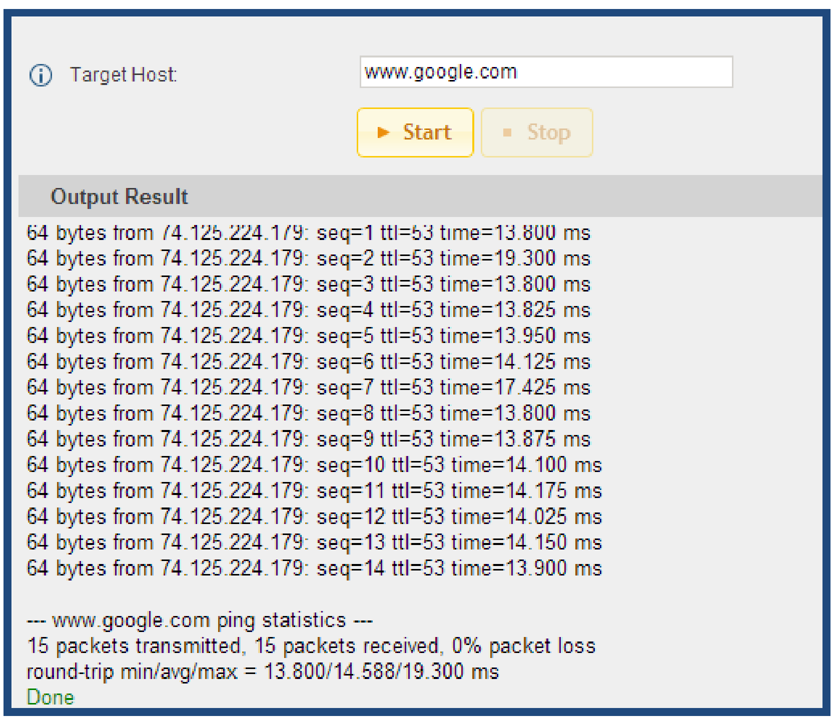


Figure 68: PING

TRACEROUTE

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

Figure 69: Traceroute

EXPERIENCING THE UCM6102/6104/6108/6116

Please visit our website: http://www.grandstream.com to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our product related documentation, FAQs and User and Developer Forum for answers to your general questions. If you have purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or submit a trouble ticket online to receive in-depth support.

Thank you again for purchasing Grandstream UCM6102/6104/6108/6116, it will be sure to bring convenience and color to both your business and personal life.

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