Grandstream Networks, Inc.

UCM6102/UCM6104/UCM6108/UCM6116

All-in-one Hybrid IPPBX Appliance

User Manual

Grandstream Networks, Inc.

www.grandstream.com

UCM6102/UCM6104/UCM6108/UCM6116 User Manual

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CHANGE LOG

This section documents significant changes from previous versions of the UCM6102/UCM6104/UCM6108/UCM6116 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 1.0.0.32

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream UCM6102/UCM6104/UCM6108/UCM6116. UCM6102/UCM6104/UCM6108/UCM6116 is an innovative, all-in-one hybrid IP PBX appliance designed for small to medium business. Powered by an advanced hardware platform with robust system resources, the UCM6102/UCM6104/UCM6108/UCM6116 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/UCM6104/UCM6108/UCM6116 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/UCM6104/UCM6108/UCM6116 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, and up to 50 SIP trunk options.
- Gigabit network port 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, 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 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
PSTN Line FXO Ports	• UCM6102: 2 ports • UCM6104: 4 ports • UCM6108: 8 ports • UCM6116: 16 ports
Network Interfaces	• UCM6108/UCM6116: Single 10M/100M/1000M RJ45 Ethernet port with integrated PoE Plug (IEEE 802.3at-2009) • UCM6102/UCM6104: 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 & 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
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS/SIP
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/UCM6104: 226mm (L) x 155mm (W) x 34.5mm (H) • UCM6108/UCM6116: 440mm (L) x 185mm (W) x 44mm (H)
Mounting	Wall mount and Desktop
Additional Features	
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT, NTT Japan (pending)
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/UCM6116: Up to 60 simultaneous calls
Conference Bridges	• UCM6102/UCM6104: Up to 3 password-protected conference bridges allowing up to 25 simultaneous PSTN or IP participants • UCM6108/UCM6116: 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 • TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 • ITU-T K.21 (Basic Level); UL 60950 (power adapter)

INSTALLATION

This section describes detailed information on installation, connection and warranty policy of the UCM6102/UCM6104/UCM6108/UCM6116.

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)

CONNECTING YOUR UCM6102/UCM6104/UCM6108/UCM6116

CONNECTING THE UCM6102

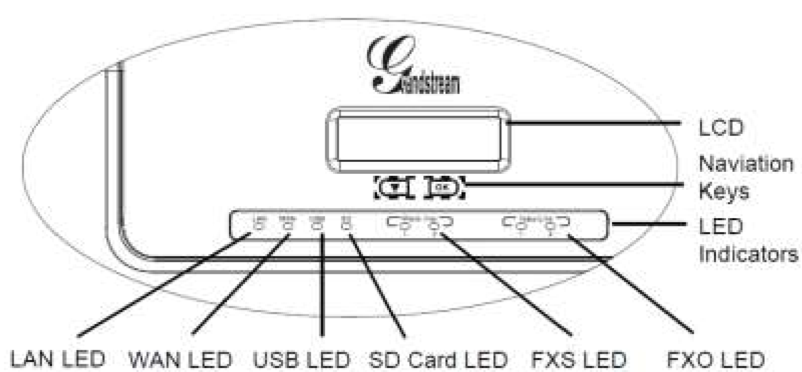


Figure 1: UCM6102 Front View

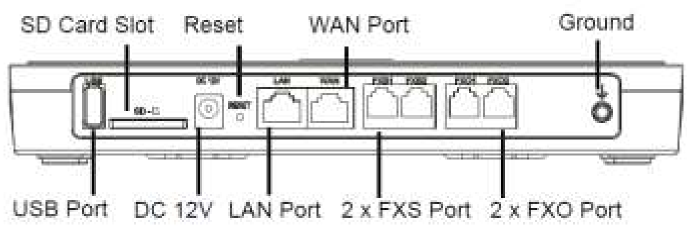


Figure 2: UCM6102 Back View

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 its 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.

CONNECTING THE UCM6104

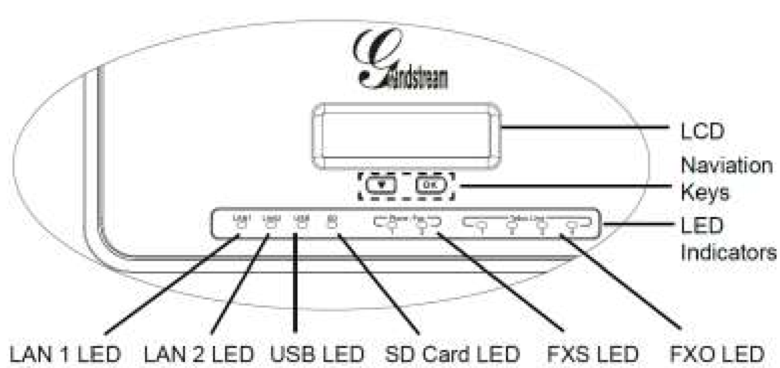


Figure 3: UCM6104 Front View

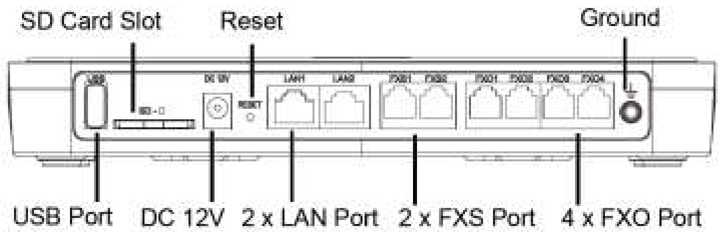


Figure 4: UCM6104 Back View

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 its 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.

CONNECTING THE UCM6108

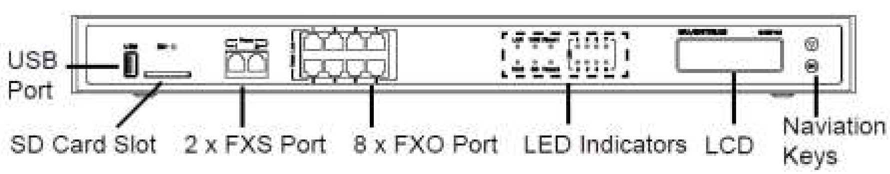


Figure 5: UCM6108 Front View

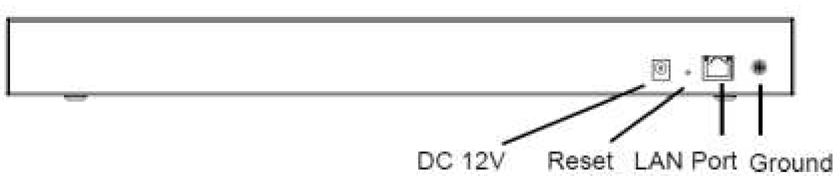


Figure 6: UCM6108 Back View

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 its 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.

CONNECTING THE UCM6116

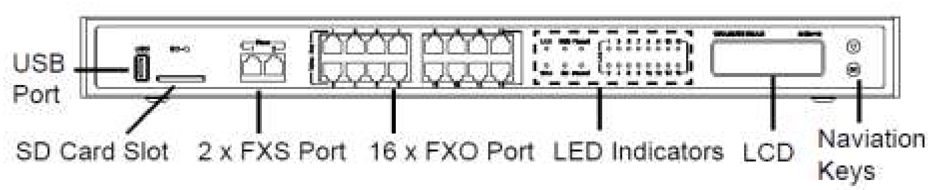


Figure 7: UCM6116 Front View

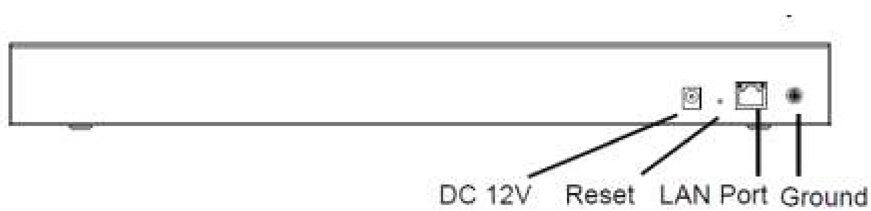


Figure 8: UCM6116 Back View

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 its 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 UCM6102/UCM6104/UCM6108/UCM6116 complies with FCC/CE and various safety standards. The UCM6102/UCM6104/UCM6108/UCM6116 power adapter is compliant with the UL standard. Use the universal power adapter provided with the UCM6102/UCM6104/UCM6108/UCM6116 package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

WARRANTY

If the UCM6102/UCM6104/UCM6108/UCM6116 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 the Grandstream Sales and Service Representative 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 UCM6102/UCM6104/UCM6108/UCM6116. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

GETTING STARTED

This section provides information about using the LCD menu, LED indicators and Web GUI of the UCM6102/UCM6104/UCM6108/UCM6116. The last section describes how to make your first call using the UCM6102/UCM6104/UCM6108/UCM6116 with your SIP phone.

USING THE LCD MENU

- **Default LCD Display**
  By default, when the device is powered on, the LCD will show device model, hardware version and IP address.
- **Menu Access**
  Press "Down" or "OK" button to start browsing menu options.
- **Menu Navigation**
  Press the "Down" arrow key to browser different menu options. Press the "OK" button to select an entry.
- **Exit**
  There is "Back" option in the menu. Select it to go back to previous menu. Also, the LCD will come back to default display after being idle in menu for more than 20 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 • **MAC**: MAC address • **Uptime**: System up time
Network Info	• **Mode**: DHCP, Static IP, or PPPoE • **IP**: IP address • **Subnet Mask**
Network Menu	• **LAN Mode**: Select LAN mode as DHCP, Static IP or PPPoE

Factory Menu	• **LCD Test Patterns**: Press "Down" button to test different LCD patterns • **Fan Mode**: Auto or On

USING THE LED INDICATORS

The UCM6102/UCM6104/UCM6108/UCM6116 has LED indicators in the front and the following table shows the status definitions.

Table 5: UCM6102/UCM6104 LED INDICATORS

LED	LED Status
LAN / WAN / FXS / FXO / USB / SD Card	Solid: Connected Flashing: Data Transferring OFF: Not Connected

Table 6: UCM6108/UCM6116 LED INDICATORS

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

USING THE WEB GUI

ACCESSING WEB GUI

The UCM6102/UCM6104/UCM6108/UCM6116 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 UCM6102/UCM6104/UCM6108/UCM6116;
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 UCM6102/UCM6104/UCM6108/UCM6116 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".

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, call routes, call features, internal options, IAX settings and SIP settings.
- **Settings:** To configure network 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.

SAVING AND APPLYING CHANGES

After configuring the web GUI options in one page, click on the "Save" button on the bottom of the page (if displayed). After saving all the changes, make sure click on "Apply Changes" button on the top right corner to submit all the changes. Follow the prompted message to reboot the device if it's required.

MAKING YOUR FIRST CALL

Power up the UCM6102/UCM6104/UCM6108/UCM6116 and your phone with network connected. Then follow the steps below to make your first call.

1. Log in the UCM6102/UCM6104/UCM6108/UCM6116 web GUI, go to **PBX**->**Basic/Call Routes**->**Extensions**;
2. Click on "Create New User" to create a new extension. You might need User ID, Password and Voicemail Password information to register and use the extension later;
3. Register the extension on your phone with the User ID, Password information;
4. When your phone is registered with the extension and ready, dial *97 to access the voicemail box. Enter the Voicemail Password and you will be prompted with the Voice Mail Main menu.
5. You are successfully connected to the PBX system now.

SYSTEM SETTINGS

This section explains configurations for system-wide parameters on the UCM6102/UCM6104/UCM6108/UCM6116. Those parameters include Network Settings, Change Password, LDAP server, HTTP server, Email settings and Time Settings.

NETWORK SETTINGS

LAN/WAN/802.1X SETTINGS

After successfully connecting the UCM6102/UCM6104/UCM6108/UCM6116 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 depending on the network environment. The settings are similar for UCM6104/UCM6108/UCM6116. The UCM6102 supports both WAN port and LAN port, with Router or Switch mode function configurable on the LAN port. Select each tab in the **Network Settings** page to configure LAN settings, WAN settings (UCM6102 only) and 802.1X.

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

Table 7: NETWORK SETTINGS

Settings -> Network Settings -> LAN	
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.
Preferred DNS Server	Enter the preferred DNS server address.
Settings -> 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.

Table 8: UCM6102 NETWORK SETTINGS

Settings -> Network Settings -> WAN	
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.
Preferred DNS Server	Enter the preferred DNS server address.
Settings -> Network Settings -> LAN	
Mode	Select LAN port mode as Router or Switch.
IP Address	Enter the IP address assigned to the LAN port.
Subnet Mask	Enter the subnet mask.
DHCP Server Enable	Enable or disable DHCP server capability.
DNS Server 1	Enter DNS server address 1.
DNS Server 2	Enter DNS server address 2.
Allow IP Address From	Enter the IP Pool starting address.
Allow IP Address To	Enter the IP Pool ending address.
Default IP Lease Time	Enter the IP lease time (in seconds).
Settings -> 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.
Settings -> 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. Up to 8 IP address can be configured.
LAN Port	Specify the LAN port number. Up to 8 ports can be configured.
Protocol Style	Select protocol type for the forwarding in the selected port.

NETWORK SECURITY SETTINGS

The UCM6102/UCM6104/UCM6108/UCM6116 provides users Firewall configurations to prevent certain malicious attack to the device system, allow, restrict or reject specific traffic through the device for security and bandwidth purpose. Go to Web GUI->**Settings**->**Network Settings**->**Security** page, users will see the current service information with Port, Process and Type, as well as Firewall settings.

Port	Process	Type
7777	asterisk	tcp/IPv4
389	slapd	tcp/IPv4
22	dropbear	tcp/IPv4
8089	lighttpd	tcp/IPv4
17210	-	udp/IPv4
69	opentftpd	udp/IPv4
6002	-	udp/IPv4
9090	asterisk	udp/IPv4
6060	zero_config	udp/IPv4
8899	-	udp/IPv4
5060	asterisk	udp/IPv4
4569	asterisk	udp/IPv4
5353	zero_config	udp/IPv4
36076	syslogd	udp/IPv4

Figure 10: Current Service

Users could configure the following options for the Firewall settings.

- Interface. Select the interface (LAN/WAN) For firewall settings.

- Ping Enable. Enable or disable ICMP response for Ping request. The default setting is Yes.
- SYN Flood. Enable to prevent SYN Flood denial-of-service attack to the device.
- Death-of-Ping. Enable to prevent Death-of-Ping attack to the device.
- Create New Rule.
  Click on "Create New Rule" button and a new window will pop up to specify rule options.

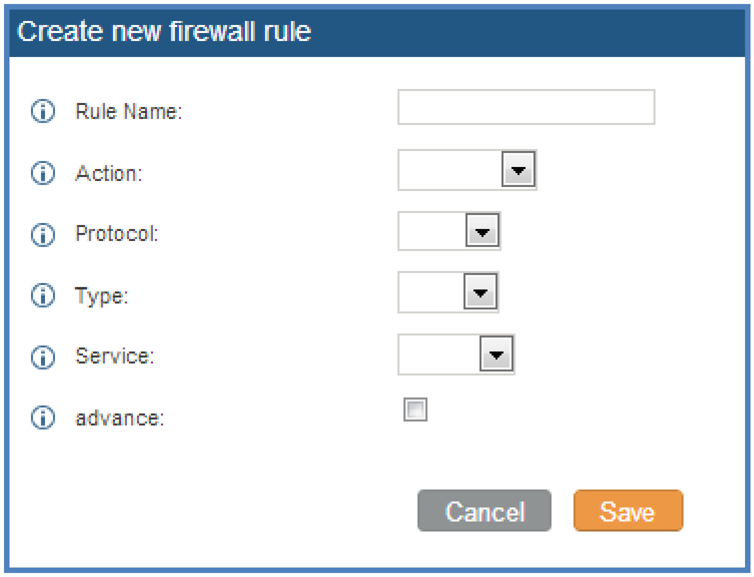


Figure 11: Create New Firewall Rule

Table 9: Firewall Rule Settings

Rule Name	Specify the Firewall rule name.
Action	Select the action for the Firewall to perform. • ACCEPT • REJECT • DROP
Protocol	Select the protocol for the traffic. • TCP • UDP • Both
Type	Select the traffic type. • IN. If selected, users will need specify the interface for the incoming packets. • OUT
Service	Select the service type. • FTP • SSH • Telnet

	• TFTP • HTTP • LDAP
Advance	Check the box to display advanced options. • Source Enter the source IP address and port. • Destination Enter the destination IP address and port.

Click on "Apply" button to save the change and then submit by clicking on "Apply Changes". The new rule will then display at the bottom of the page. Users can select to edit the rule, or select to delete the rule.

CHANGE PASSWORD

After login the Web GUI for the first time, it is highly recommended for users to change the default password "admin" to 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 UCM6102/UCM6104/UCM6108/UCM6116 has an embedded LDAP server for users to manage corporate phonebook in a centralized way. By default, the LDAP server has generated the phonebook based on the created extensions already. If users have the Grandstream phone provisioned by the UCM6102/UCM6104/UCM6108/UCM6116, the LDAP directory has been set up on the phone and can be used right away. Or 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

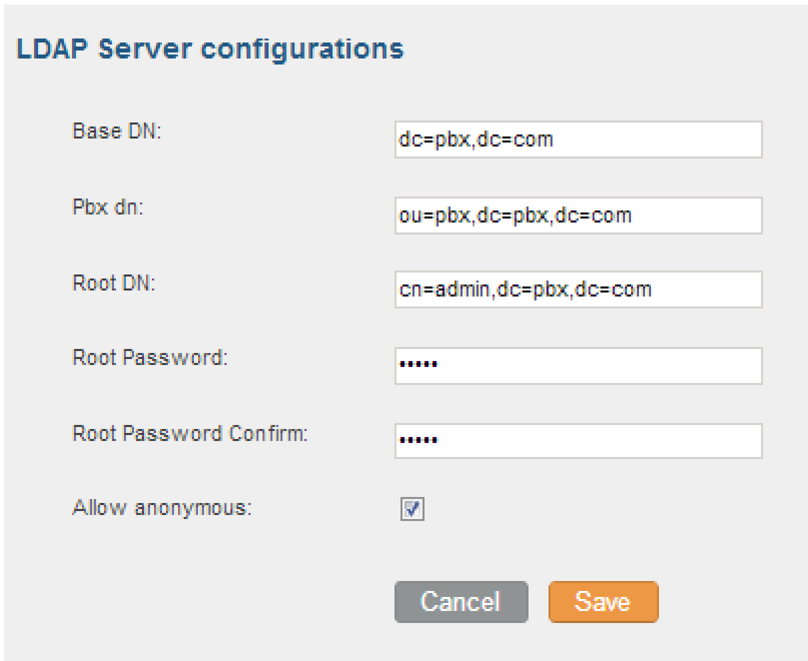


Figure 12: LDAP Server Configurations

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

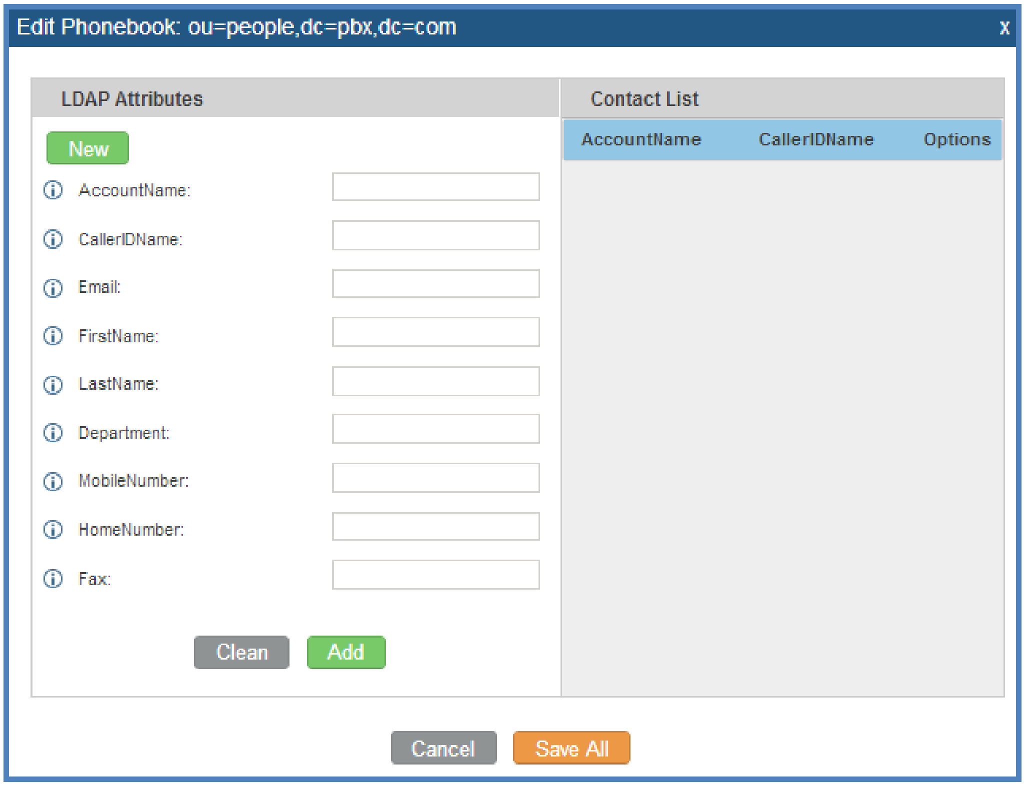


Figure 13: Add New LDAP Phonebook

LDAP CLIENT CONFIGURATIONS

To configure the LDAP client so the default PBX phonebook can be used, follow the instructions in the LDAP Client Configuration section.

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 GXP2200 to successfully use the LDAP server as configured in ***Figure 12: LDAP Server Configurations***.

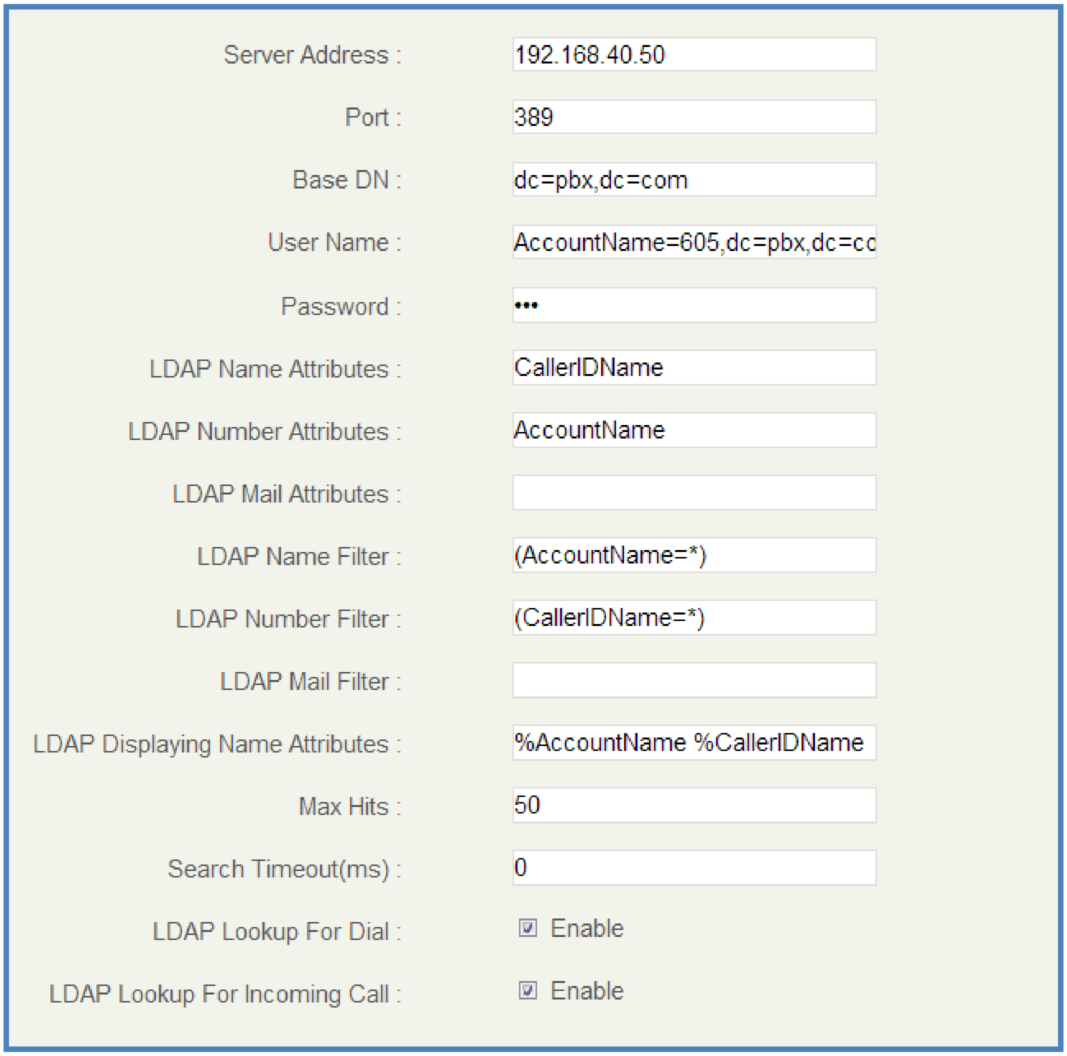


Figure 14: GXP2200 LDAP Phonebook Configuration

HTTP SERVER

The UCM6102/UCM6104/UCM6108/UCM6116 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 10: 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.

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 UCM6102/UCM6104/UCM6108/UCM6116 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 11: 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.

TIME SETTINGS

The current system time on UCM6102/UCM6104/UCM6108/UCM6116 can be checked under Web GUI->**Status**->**System Status**. To change the time settings, go to Web GUI->**Settings**->**Time Settings**.

Table 12: Time Settings

NTP Server	Specify the URL or IP address of the NTP server for the PBX to synchronize the date and time. The default NTP server is ntp.ipvideotalk.com.
Enable DHCP Option 2	If set to "Yes", the device 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 device is allowed to get provisioned for NTP Server from DHCP Option 42 in the local server automatically. This will then override the NTP Server manually configured on the PBX. The default setting is "Yes".
Time Zone	Select the proper time zone option so the PBX can display correct date and time accordingly. If "Automatic" is selected, the PBX will obtain the time zone information according to the detected IP location.
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 with 1 hour ahead which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it 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…) 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 or HTTP/HTTPS download. All Grandstream SIP devices support a proprietary binary format configuration file as well as XML format configuration file. The UCM6102/UCM6104/UCM6108/UCM6116 provides a Plug and Play mechanism to auto-provision the Grandstream SIP devices in a zero configuration manner, which 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 UCM6102/UCM6104/UCM6108/UCM6116 is capable to accomplish the following configurations on the SIP end point device.

- Assign an extension to the phone.
- Set up config server download path for further provisioning purpose.
- Set up LDAP client side configurations to use the PBX default phonebook.

This section explains how zero config works on the UCM6102/UCM6104/UCM6108/UCM6116. 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 enabled on the UCM6102/6104/6108/6116 for auto provisioning. Three methods of auto provisioning are used (see below).

- **SIP SUBSCRIBE**

  The UCM6102/UCM6104/UCM6108/UCM6116 can automatically discover the phones in the same network using PnP feature with multicast SUBSCRIBE/NOTIFY. All current Grandstream phones support PnP feature and will send SUBSCRIBE at boot up and in the process be discovered by the PBX with the same type PnP feature support.

  On the phone side, after the phone boots up, it will send out multicast SUBSCRIBE message. When the UCM6102/UCM6104/UCM6108/UCM6116 receives the SUBSCRIBE, a SIP NOTIFY message will be sent to the phone with config server path URL in the NOTIFY message body. The phone will then use the path to download the config file generated in the UCM6102/UCM6104/UCM6108/UCM6116.

- **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. The phone will then use the path to download the config file generated in the UCM6102/UCM6104/UCM6108/UCM6116.

- **mDNS**

  The mDNS process is similar to the SIP SUBSCRIBE.

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

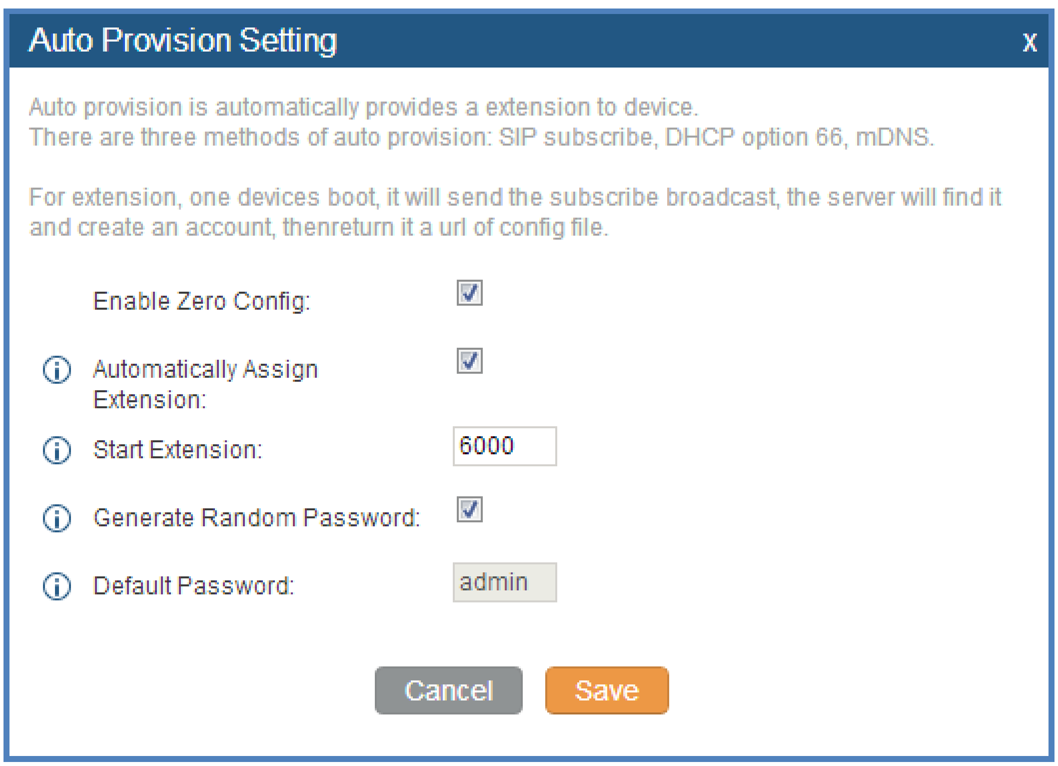


Figure 15: Auto Provision Setting

Table 13: Auto Provision Setting

Enable Zero Config	Enable or disable the zero config feature on the PBX. The default setting is Yes.
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" to start the discovery and provisioning process. Reboot the device and the assigned extension will be registered after booting up.

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. Then click on "Save" to start discovering the devices within the same network.

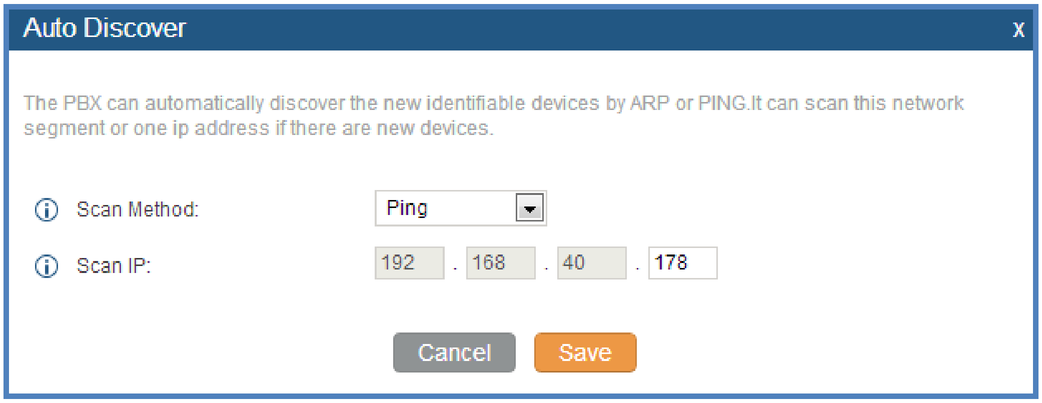


Figure 16: Auto Discover

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.

No.	Mac Address	IP Address	Extension	Version	Vendor	Model	Connect State	Create Config	Options
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 17: Discovered Devices

ASSIGNMENT

In the discovered list, click on  to assign an extension to the device.

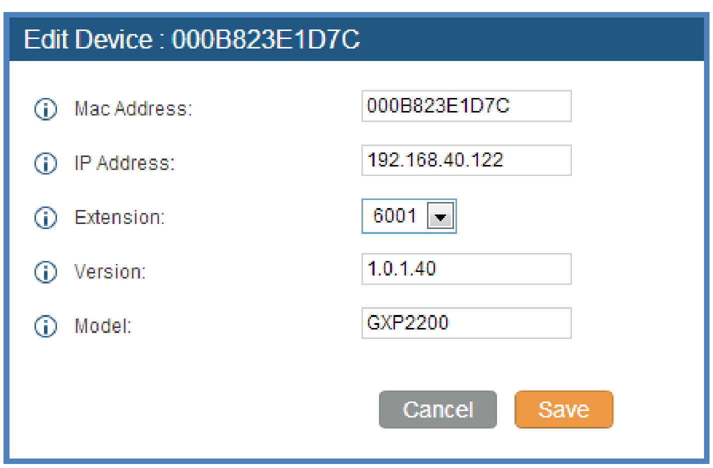


Figure 18: Assign Extension To Device

Users could also directly create a new device and assign the extension at one time. Click on "Create New Device" and the following window will be popped out. 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 19: 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:

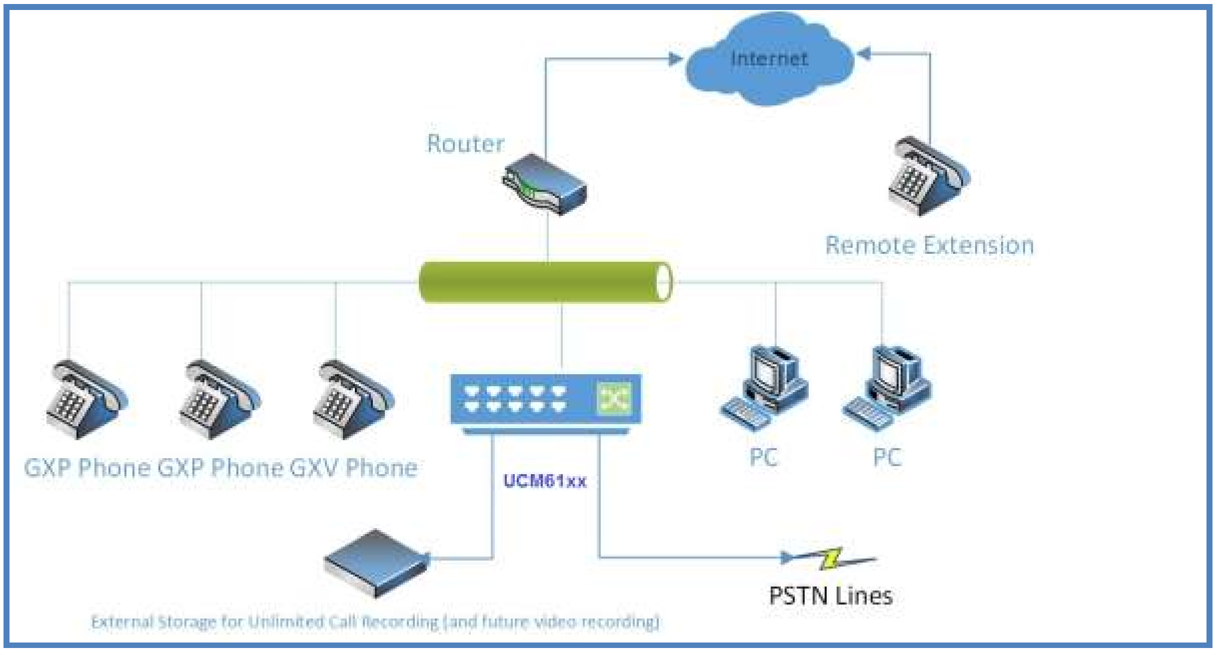


Figure 20: Provisioning Example 1

The above figure shows a common setup among small businesses, where the UCM6102/UCM6104/UCM6108/UCM6116 is placed behind a company's router or firewall. The phones are in the same network as the UCM6102/UCM6104/UCM6108/UCM6116 and can be discovered automatically by UCM6102/UCM6104/UCM6108/UCM6116 using the Zero Config feature.

Example 2:

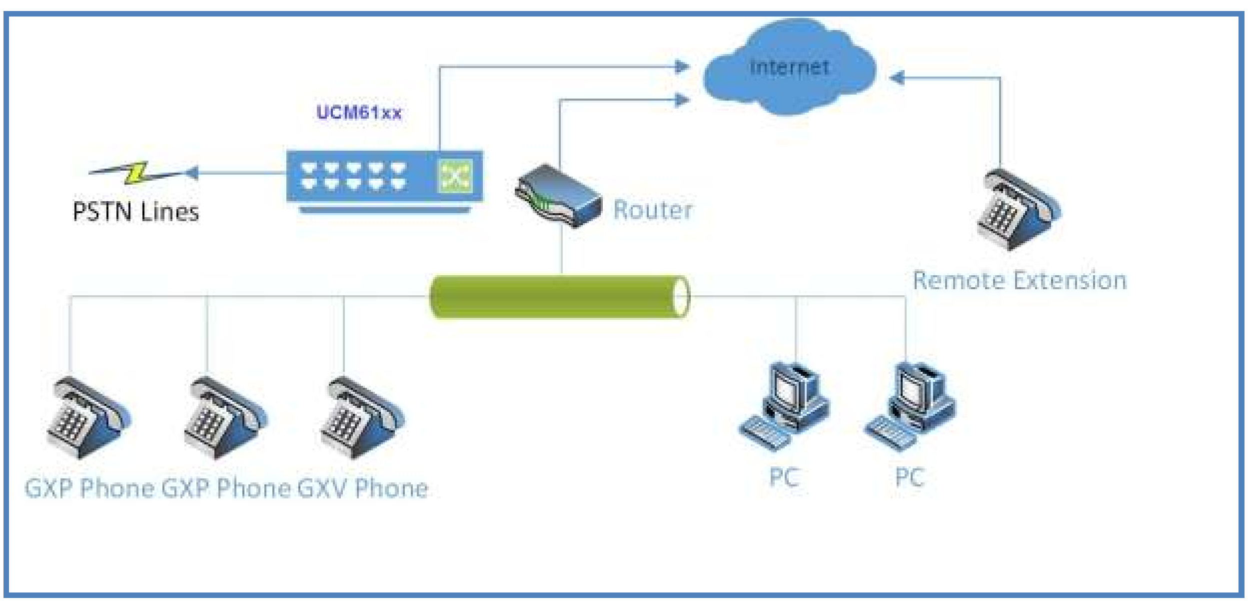


Figure 21: Provisioning Example 2

This is another typical setup. In this setup, the UCM6102/UCM6104/UCM6108/UCM6116 is placed directly over the internet (outside from the network where the phones are deployed). Under this topology, the UCM6102/UCM6104/UCM6108/UCM6116 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 UCM6102/UCM6104/UCM6108/UCM6116 will need help to get the phones to point itself to the UCM6102/UCM6104/UCM6108/UCM6116 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 UCM6102/UCM6104/UCM6108/UCM6116 and send request for config file.
- When the phone requests cfgMAC.xml from the UCM6102/UCM6104/UCM6108/UCM6116, the UCM6102/UCM6104/UCM6108/UCM6116 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 window will show to fill in the details. The configuration parameters are as follows.

Table 14: 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	Select permission for the user. The available permissions are Internal, Local, National and International. The default permission is Internal.
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 enabled.
Voicemail Password	Configure Voicemail password (digits only). A random numeric password is automatically generated.
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 sending to the user's voicemail (if enabled) or hangup. The default setting is 60 seconds.
Technology	

SIP	Check SIP if the user is using SIP or a SIP device.
IAX	Check IAX if the user is using IAX or a IAX device.
Analog Station	Select the port number if the user is attached on the analog port of the PBX.
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.
Call 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 matching peers by IP address without matching port number. • Invite: Authentication of incoming INVITE messages is not required. • No: Normal IP-based matching and authenticated INVITES. The default setting is Port.
Enable Keep-alive	If enabled, keep the NAT session open. The default setting is enabled.
Keep-alive Frequency	Configure the number of seconds for the host to be up for Keep-alive.
Other Settings	
SRTP	Enable SRTP for the call.
FAX Detect	Enable to detect fax signal from the user/trunk during the call and send the received fax to the Email address configured in this configuration page. If no Email address can be found for the user, send the received fax to the default Email address in FAX setting page. **Note:** If enabled, FAX cannot use Passthrough.
Strategy	This option controls how the extension can be used on the device. • Allow all Device in any network can register using the extension.

	• Only local subnets Only the user in specific subnet can register using the extension. Up to three subnet can be specified. • A specific IP Address. Only the device on the specific IP address can register using the extension. The default setting is "Allow all".
Disable Password	If set to Yes, when dialing out with outgoing rules, the user doesn't need enter the password.
Codec Preference	Select audio and video codec for the user. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, 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 15: Batch Add Extension Parameters

General	
Start Extension	The starting extension number.
Create Number	The number of extensions to be added.
Permission	Select permission for the user. The available permissions are Internal, Local, National and International. The default permission is Internal.
Enable Voicemail	Enable Voicemail for the user. The default setting is enabled.
SIP/IAX Password	Configure the SIP/IAX password for the users. • 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.
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.

Technology	
SIP	Check SIP if the users are using SIP or a SIP device.
IAX	Check 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.
Call 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 matching peers by IP address without matching port number. • Invite: Authentication of incoming INVITE messages is not required. • No: Normal IP-based matching and authenticated INVITES. The default setting is Port.
Enable Keep-alive	If enabled, keep the NAT session open. The default setting is enabled.
Keep-alive Frequency	Configure the number of seconds for the host to be up for Keep-alive.
IAX Settings	
Max Call Numbers	Limit the maximum amount of call numbers allowed for a single IP address.
Require Call Token	If set to Yes, call token is required. If set to "Auto", it may 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.
FAX Detect	Enable to detect fax signal from the user/trunk during the call and send the received fax to the Email address configured in this configuration page. If no Email address can be found for the user, send the received

	fax to the default Email address in FAX setting page. **Note:** If enabled, FAX cannot use Passthrough.
Disable Password	If set to Yes, when dialing out with outgoing rules, the user doesn't need enter the password.
Codec Preference	Select audio and video codec for the user. The available codecs are: PCMU, PCMA, GSM, G.726, G.722, G.729, G.723, ILBC, ADPCM, LPC10, H.264, H.263, H.263p.

EDIT EXTENSION

All the PBX extensions are listed under Web GUI->**PBX**->**Basic/Call Routes**->**Extensions**, with CallerID Name, Technology, IP/Port and registration status displayed. Each extension has a checkbox to be selected and options for users to edit.

- **Edit single extension**

  Click on to start editing the extension. The configuration options are listed in ***Table 14: Extension Configuration Parameters***.

- **Reboot the user**

  Click on to send NOTIFY reboot event to the device with the extension registered.

- **Delete single extension**

  Click on to delete the extension.

- **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 15: 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.

The analog trunk options are listed in the table below.

Table 16: 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	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. Enable to turn this feature on
Busy Count	If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results can be achieved if set to 6 or even 8. Mind that the higher the number, the more time that will be needed to hangup a channel, but lowers the probability that you will get random hangups.",
Congestion Detection	Congestion detection is used to detect far end congestion signal. Enable to turn this feature on.
Congestion Count	If congestion detection is enabled, it is also possible to specify how many congestion tones to wait for. The default setting is 2.
Enable Polarity Reversal	If this option is enabled, the reception of a polarity reversal will mark when a outgoing call is answered by the remote party. in some countries, a polarity reversal is used to signal the disconnect of a phone line, the call will be considered \"hung up\" on a polarity reversal.",
Polarity On Answer Delay	minimal time period (ms) between the answer polarity switch and

	hangup polarity switch. (default: 600ms)",
RX Gain	Gain for the receive channel of analog FXO port. Range: -13.5 (dB) to + 12.0 (dB).
TX Gain	Gain for the transmit channel of analog FXO port. Range: -13.5 (dB) to + 12.0 (dB).
Ring Timeout	unit: milisecond . Trunk (FXO) devices must have a timeout to determine if there was a hangup before the line was answered. This value can be tweaked to shorten how long it takes before asterisk considers a non-ringing line to have hungup.",
Use CallerID	Enabling this option enabled Caller ID detection
Caller ID Start	This options allows one to define the start of a Caller ID signal: Ring, to start when a ring is received, or Polarity, to start when a polarity reversal is started
Current Disconnect Threshold (ms)	This is the periodic time in milliseconds that the PBX will use to check on a voltage drop in the line. Default setting is 200ms.
CID Signalling	This option defines the type of Caller ID signalling to use: bell (bell202 as used in the United States), v23 (as used in the UK), v23_jp (as used in Japan), or dtmf (as used in Denmark, Sweden, and Holland)."
Tone Country	Select country for tone settings. You can also select Custom and set the values manually.
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 edit manually. Default value: f1=480,f2=620,c=250/250
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 edit manually.

	Default value: f1=480,f2=620,c=250/250

VOIP TRUNKS

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

- Click on "Create New SIP/IAX Trunk" to add a new VoIP trunk first. Then click on to configure more options for the VoIP trunk.
- Click on to delete the VoIP trunk.

The VoIP trunk options are listed in the table below.

Table 17: 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	A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc.",
Host Name	IP address or URL for your VoIP providers server.",
VoIP Trunk Configuration Parameters	
Provider Name	Configure the provider name for the VoIP trunk.
Host Name	Configure the host name for the VoIP trunk.
Transport	Configure the SIP transport protocol "UDP", "TCP" or "TLS".
Caller ID	Configure the Caller ID.
CallerID Name	The new name of caller to replace when extension was not set CallerID Name.",
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 PBX will send SIP OPTIONS to check if the device is still alive. The default setting is disabled.
Enable FAX Detect	Enable both CNG and T.38 detect. The default setting is disabled.

SRTP	Enable SRTP for the VoIP trunk. The default setting is disabled.

CALL ROUTES

OUTBOUND ROUTES

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 sequence.

Table 18: Outbound Route Configuration Parameters

Calling Rule Name	Configure the calling rule name (e.g., local).
Pattern	• All patterns are prefixed with the "_". • 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. • Local: The lowest level required. All users can use this rule. • National: Users with National level 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.
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  to edit the inbound route.

- Click on 🗑 to delete the inbound route.

Table 19: Inbound Route Configuration Parameters

Trunks	Select the trunk to configure the inbound route.
DID Pattern	• All patterns are prefixed with the "_". • 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. • Local: The lowest level required. All users can use this rule. • National: Users with National level 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. • Extension • Extension's voicemail • Call Queue • Conference Room • Operator • Hangup • Voicemail Dial Code • Congestion • Local Extension 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.
Destination	Select the destination when the inbound rule is used at the configured time range.. • Extension • Extension's voicemail

	• Call Queue • Conference Room • Operator • Hangup • Voicemail Dial Code • Congestion • Local Extension by DID
DID Features	
Dial Trunk	If enabled, users can dial outbound calls by DID through inbound trunks. The privilege level can be set according to the corresponding inbound rules.
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.

CONFERENCE BRIDGE

Conference bridge configurations can be accessed under Web GUI->**PBX**->**Call Features**->**Conference**. Users could create, edit, view and delete conference bridges. The conference room status and activity will show in the page as well.

- Click on "Create New Conference Room" to add a new conference bridge.
- Click on to edit the conference room.
- Click on to invite a user to the conference. The user will receive the ring to join the conference.
- Click on to kick a participant in the conference. This will hang up the conference call on the user.
- Click on to lock the conference room.
- Click on to delete the conference bridge.

Table 20: Conference Bridge Configuration Parameters

Extension	Configure the conference number for the users to dial and join the conference.
Password	When configured, the users who would like to join the conference call must enter this password before accessing the conference room.
Admin Password	Configure the password to join the conference room as administrator.
Enable Caller Menu	When enabled, conference participant could press the * key to access the conference bridge menu. The default setting is disabled.
Record conference	When enabled, the calls in this conference room will be recorded in a .wav format file. The default setting is disabled.
Quite Mode	When enabled, if there are users entering or leaving the conference room, voice prompt or notification tone won't be played. The default setting is disabled.
Wait For Admin	When enabled, the participants will not hear each other until the admin joins the conference room. The default setting is disabled.
Enable User Invite	When enabled, users could press 0 to invite other users to join the conference. The default setting is disabled. **Note:**

	Admin can always press 0 to invite other users to the conference.
Announce Callers	When enabled, announcement will be made to all conference participants when there is user joining in the conference. The default setting is disabled.
Play Hold Music For First Caller	When enabled, the PBX will play Hold music to the first participant until another user joins the conference room. The default setting is disabled.
Skip Authentication When Invite User via Trunk from Web GUI	When enabled, the invitation from Web GUI for the conference bridge with password will skip the authentication for the invited users. Please be cautious to use this option. The default setting is disabled.

IVR

CONFIGURING IVR

IVR configurations can be accessed under Web GUI->**PBX**->**Call Features**->**IVR**. Users could create, edit, view and delete 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 21: IVR Configuration Parameters

Name	Configure the name of the IVR. Letters, digits, underscore and hyphen are allowed.
Extension	Enter the extension number for users to access the IVR.
Dial Other Extensions	If enabled, callers to the IVR can dial extensions other than the ones explicitly defined in the IVR. The default setting is disabled.
Dial Trunk	If enabled, users can use trunk with configured permission.
Permission	Select permission for users to use trunk for outgoing calls.
Welcome Prompt	Select a audio file to play as the welcome prompt. Click on "Prompt" to add additional audio file.
Timeout	After playing the prompts in the IVR, the PBX will wait a period of time to detect DTMF entry. This period of time is the timeout (in seconds). 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	Select the repeat time if no DTMF input. After the repeat time, go to the timeout destination if configured, otherwise hangup. The default setting is 4.
Invalid Repeat Loops	Select the repeat time if the input is invalid. After the repeat time, go to the timeout destination if configured, otherwise hangup. The default setting is 4.
Key Press Event	Select the event for each key pressing for 0-9, *.

	The event options are: Extension VoiceMail Conference Rooms VoiceMail Group IVR Ring Group Queues Page Group IVR Prompt Hangup

CREATING IVR PROMPT

To Record New IVR Prompt or Upload IVR Prompt, 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.

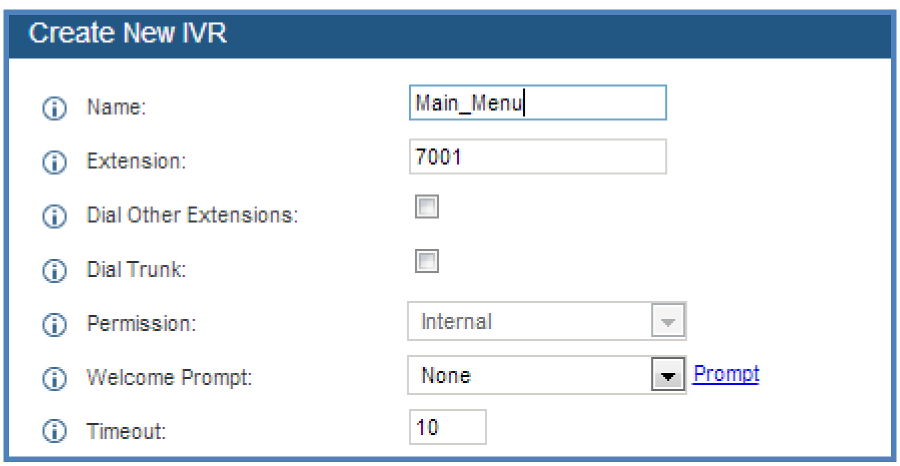


Figure 22: Click On Prompt To Create IVR Prompt

RECORD NEW IVR PROMPT

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

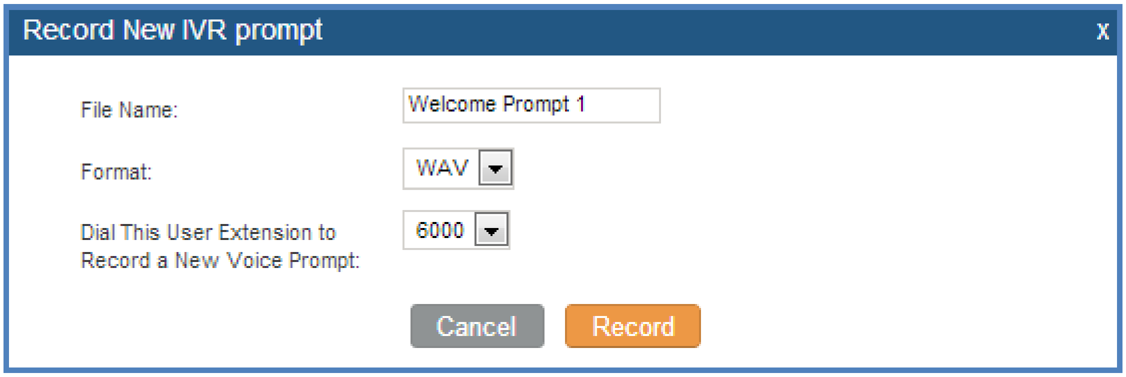


Figure 23: Record New IVR Prompt

- Specify the IVR file name.
- Select the format (GSM or WAV) for the IVR file.
- Select the extension which will be dialed for the user to start recording the voice prompt.
- Click the "Record" button. A request will be sent to the PBX and the PBX will then call the extension for recording.
- Pick up the call from the extension and start the recording.

UPLOAD IVR PROMPT

In Web GUI->**PBX**->**Internal Options**->**IVR Prompt** page, click on "Upload IVR Prompt" and choose a file to upload. The requirement on the audio file is as follows:

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

Click on to select audio file from local PC and click on to start uploading.

VOICE PROMPT

Language settings for voice prompt can be accessed under Web GUI->**PBX**->**Internal Options**->**Language**. Users could upload a voice prompt package and then select the language in the list with available language options.

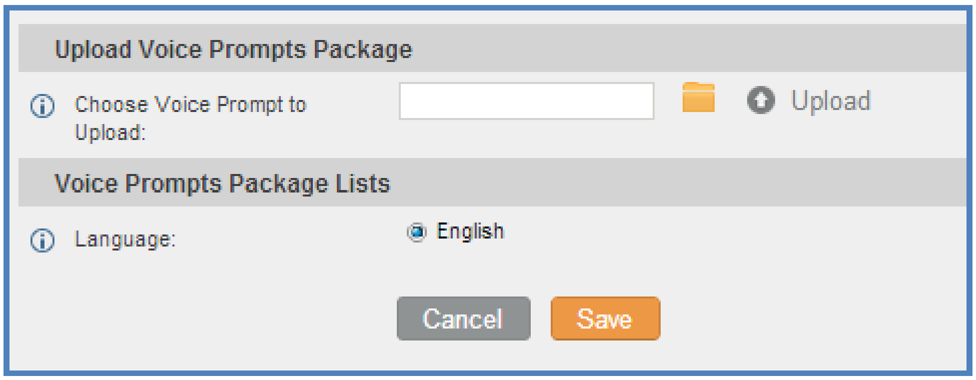


Figure 24: Language Settings For Voice Prompt

- Click on the folder icon to select a voice prompt package from local PC.

  The uploaded file must be smaller than 20 megabytes with package structure:
  [Package]
  [voice prompt dir] | [... dir] | [... files]
  info.txt (containing the language name for display, in UTF8)

- Click on the upload icon to start uploading.
- Voice Prompts Package Lists allows users to select the default language for voice prompt for internal calls, inbound calls and outbound calls.

VOICEMAIL

CONFIGURING VOICEMAIL

General Voicemail settings can be accessed via Web GUI->**PBX**->**Call Features**->**Voicemail**.

Users could configure the PBX to send the users Email with the voicemail as attachment. Click on "Email Settings For Voicemails" button to configure the Email attributes and content.

Table 22: Email Settings For Voicemails

Attach Recordings to E-Mail	If enabled, voicemails will be sent to user's Email address as the configured template. The default setting is enabled.
Template For Voicemail Emails	Fill in the "From:", "Subject:" and "Message:" content. The template variables are: • \t: TAB • ${VM_NAME}: Recipient's firstname and lastname • ${VM_DUR}: The duration of the voicemail message • ${VM_MAILBOX}: The recipient's extension • ${VM_CALLERID}: The caller id of the person who left the message • ${VM_MSGNUM}: The message number in your mailbox • ${VM_DATE}: The date and time the message was left

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

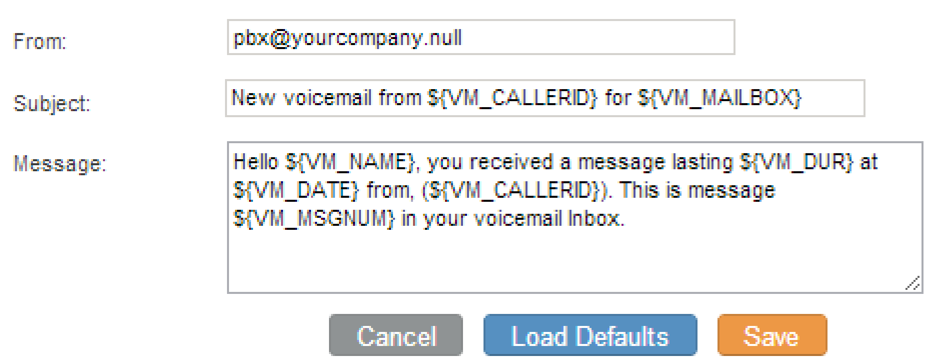


Figure 25: Default Email Template

Table 23: 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.
Max Messages Per Folder	Configure the maximum number of messages in users' voicemail folders. The default setting is 25.
Max Message Time	Configure the maximum length of the voicemail message (in seconds). The default setting is 2 minutes.
Say Message Caller-ID	If enabled, the caller ID of the user leaving the message will be announced at the beginning of the voicemail message. The default setting is disabled.
Say Message Duration	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is disabled.
Play Envelope	If enabled, introduction of each message will be played when accessed from the voicemail application. The default setting is enabled.
Allow Users To Review	If enabled, users can review the message before sending out. The default setting is disabled.

CONFIGURING VOICEMAIL GROUP

Voicemail Group can be configured under Web GUI->**PBX**->**Call Features**->**Voicemail Group**. In this page, users could create voicemail group which contains members that will receive the voicemail if the voicemail group extension has voice messages. Click on "Create New Voicemail Mail Group" to configure the group.

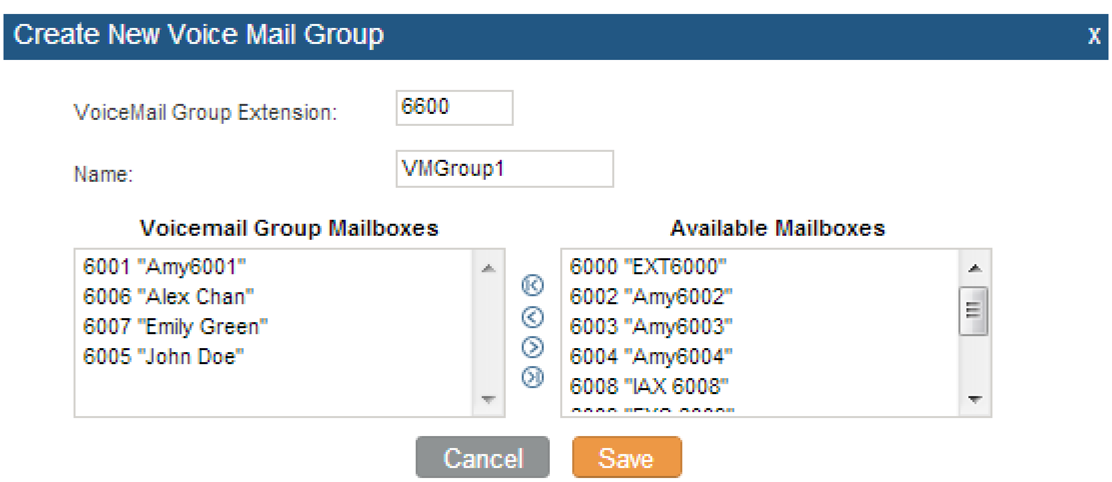


Figure 26: Voicemail Group

- Enter the Voicemail Group Extension. The voicemail messages left to this extension will be forwarded to all the voicemail group members.
- Configure the Name to identify the voicemail group. Letters, digits, underscore and hyphen are allowed.
- Select available mailboxes from the right list and add them to the left list.
- Click Save the finish the configuration.

RING GROUP

Users could create extension for ring group which contains members that will receive the call with specific ring strategy if the group extension has incoming calls.

CONFIGURING RING GROUP

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

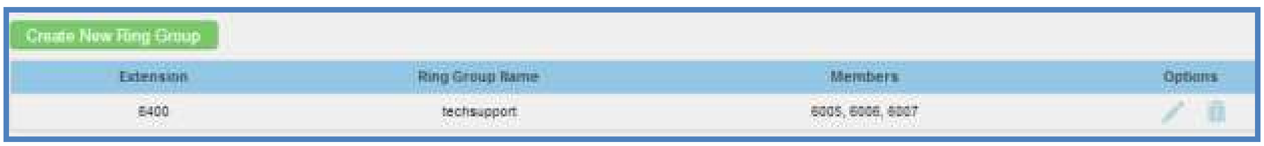


Figure 27: Ring Group

- Click on "Create New Ring Group" to add ring group.
- Click on to edit the ring group.
- Click on to delete the ring group.

RING GROUP PARAMETERS

Table 24: Ring Group Parameters

Ring Group Name	Configure ring group name to identify the ring group. Letters, digits, underscore and hyphen are allowed.
Extension	Configure the ring group extension.
Ring Group Members	Select available users to the ring group member list. Click on to arrange the order.
Ring Strategy	Select the ring strategy. • Ring All 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 configured order one by one. If the first member doesn't answer the call, it will stop ringing the first member and start ringing the second member.

Seconds to Ring Each Member	Configure the number of seconds to ring each member. If set to 0, it will keep ringing (users could configure the ring timeout on the phone side as well). The default setting is 30 seconds.
Enable Voicemail	If enabled, the ring group extension can use voicemail.
Secret	Configure the password to access the ring group voicemail.
Email Address	Configure the Email address of the ring group extension.

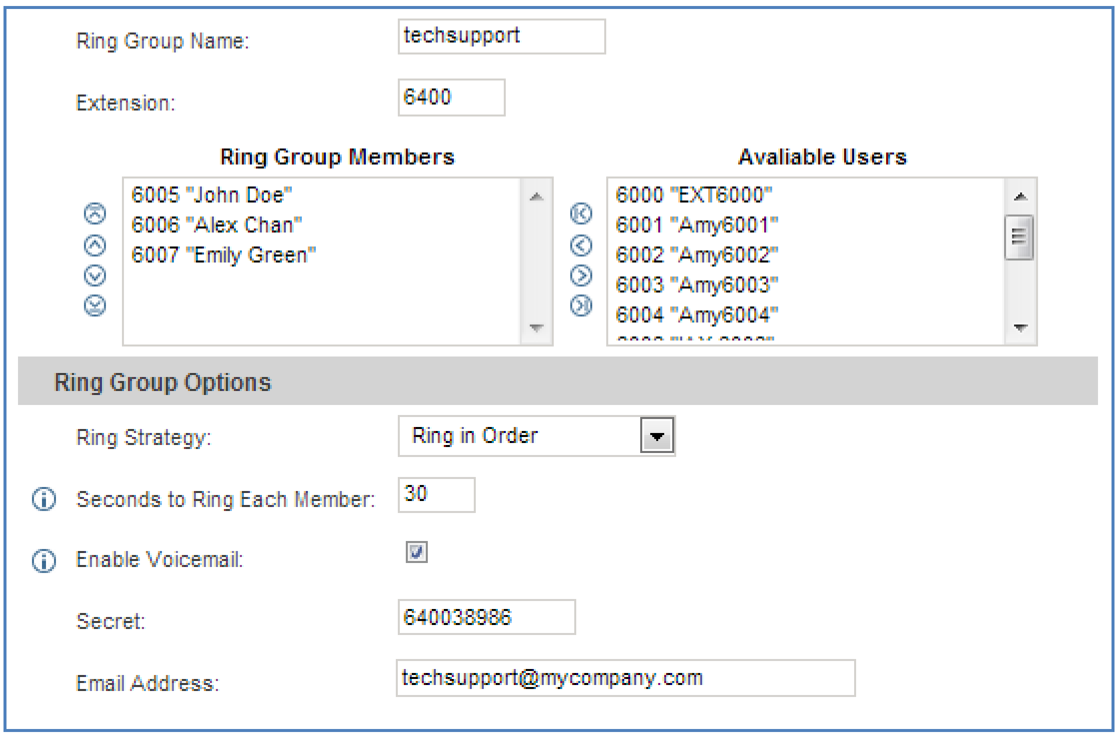


Figure 28: Ring Group Configuration

PAGING AND INTERCOM GROUP

Paging and intercom can be configured in group level under Web GUI->**PBX**->**Call Features**->**Paging/Intercom**.

- Click on "Create New Page/Intercom Group" to add page/intercom group.

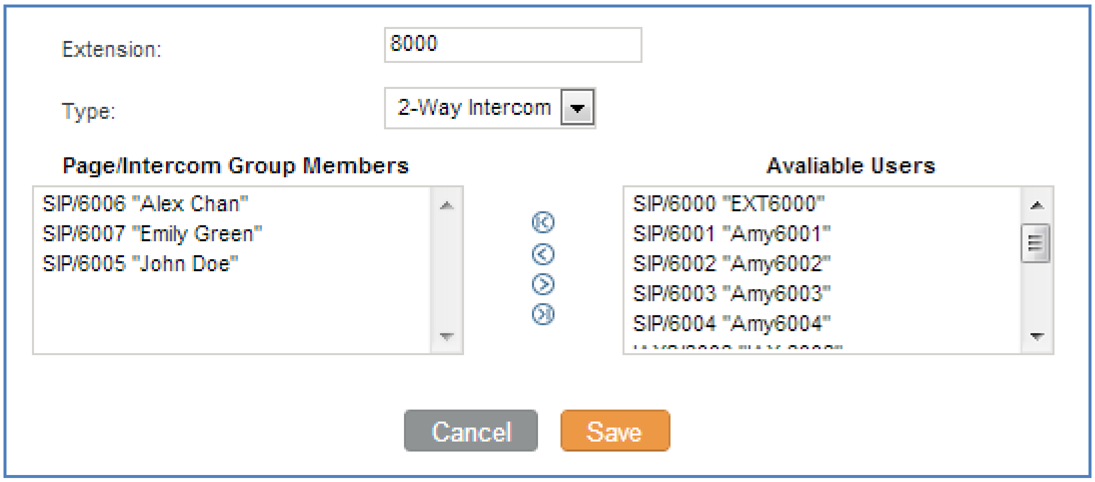


Figure 29: Page/Intercom Group

Table 25: Page/Intercom Group 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 list to 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 (see figure below). To edit page/intercom feature code, click on "Feature Codes" in the following figure.

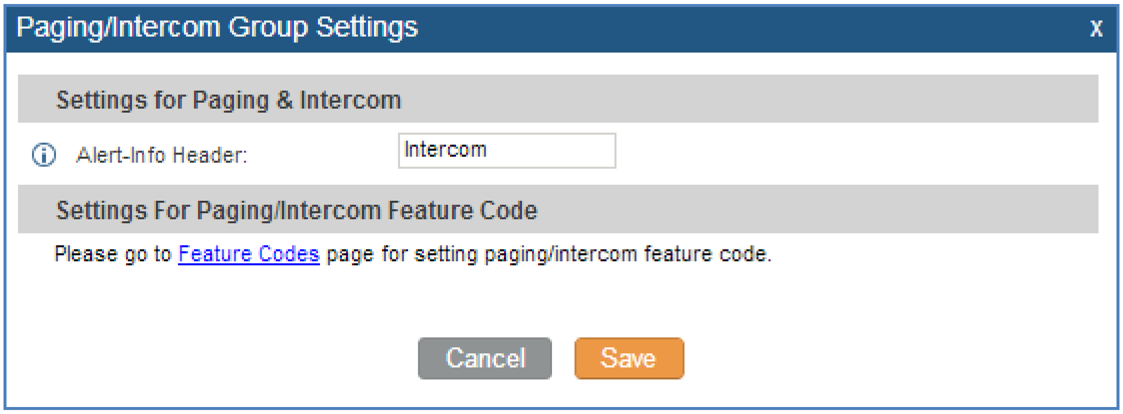
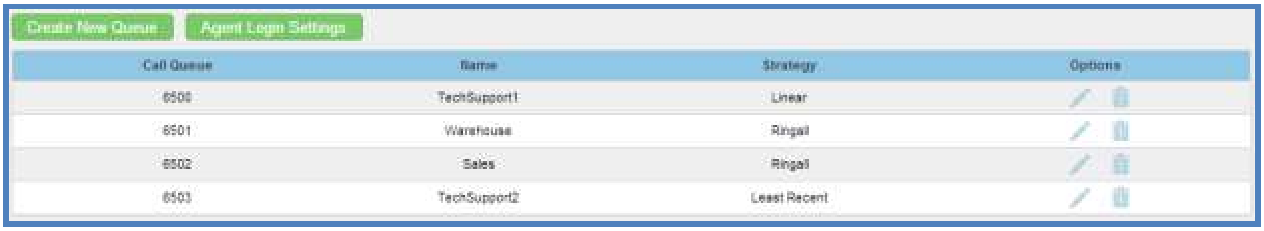


Figure 30: Page/Intercom Group Settings

CALL QUEUE

CONFIGURING CALL QUEUE

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



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

Figure 31: Call Queue

- Click on "Create New Queue" to add call queue.
- Click on to edit the call queue.
- Click on to delete the call queue.
- Click on "Agent Login Settings" to configure Agent Login Extension Postfix and Agent Logout Extension Postfix. 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 and dial 6500** to logout.

CALL QUEUE PARAMETERS

Table 26: Call Queue Parameters

Extension	Configure the call queue extension.
Name	Configure the name to identify the call queue.
Strategy	Select the strategy for the call queue. • RingAll: Ring all available Agents simultaneously until one answers. • Linear: Ring agents in the specified order specified. • LeastRecent: Ring the agent being least recent called. • FewestCalls: Ring the agent with the fewest completed calls. • Random: Ring a random agent. • RRmemory: Ring the agents in Round Robin scheduling with memory.
Music On Hold	Select the Music On Hold class for the call queue.

	Note: Music On Hold classes can be managed from **PBX**->**Internal Options**->**Music On Hold**.
Leave When Empty	Configure whether forcing the caller to leave if the call queue has no agent. • Yes: Callers will be forced to leave the call queue if the call queue is empty. • No: Never force the callers to leave the call queue when the queue is empty. • Strict: Callers will be forced to leave the call queue if the agents are paused, invalid or unavailable. This is the default setting.
John Empty	Configure whether the callers can join the call queue if the queue has no agent. The default setting is No. • Yes: Callers can always join a call queue. • No: Callers cannot join the queue if the queue has no agent. • Strict: Callers cannot join a queue if the agents are paused, in an invalid state or unavailable.
Dynamic Login Password	If enabled, the configured PIN number is required for the agent to login. The default setting is disabled.
TimeOut	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	How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. 0 means no delay."
Max Len	How many calls can be queued at once. This count does not include calls that have been connected with Agents, it only includes calls that have not yet been connected. Default is 0, which is no limit. When the limit has been reached, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue.",
Report Hold Time	If enabled, the PBX will report (to the agent) the duration of time of the call before connected to the agent.
Wait Time	If enabled, users will be disconnected after the configured number of seconds. The default setting is disabled. **Note:** It is recommended to configure Wait Time longer than the Wrapup Time.
Static Agents	Select the available agents from the right list to the left. Click on to arrange the order.

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 the music files. The "default" Music On Hold class already have 5 audio files defined for users to use.

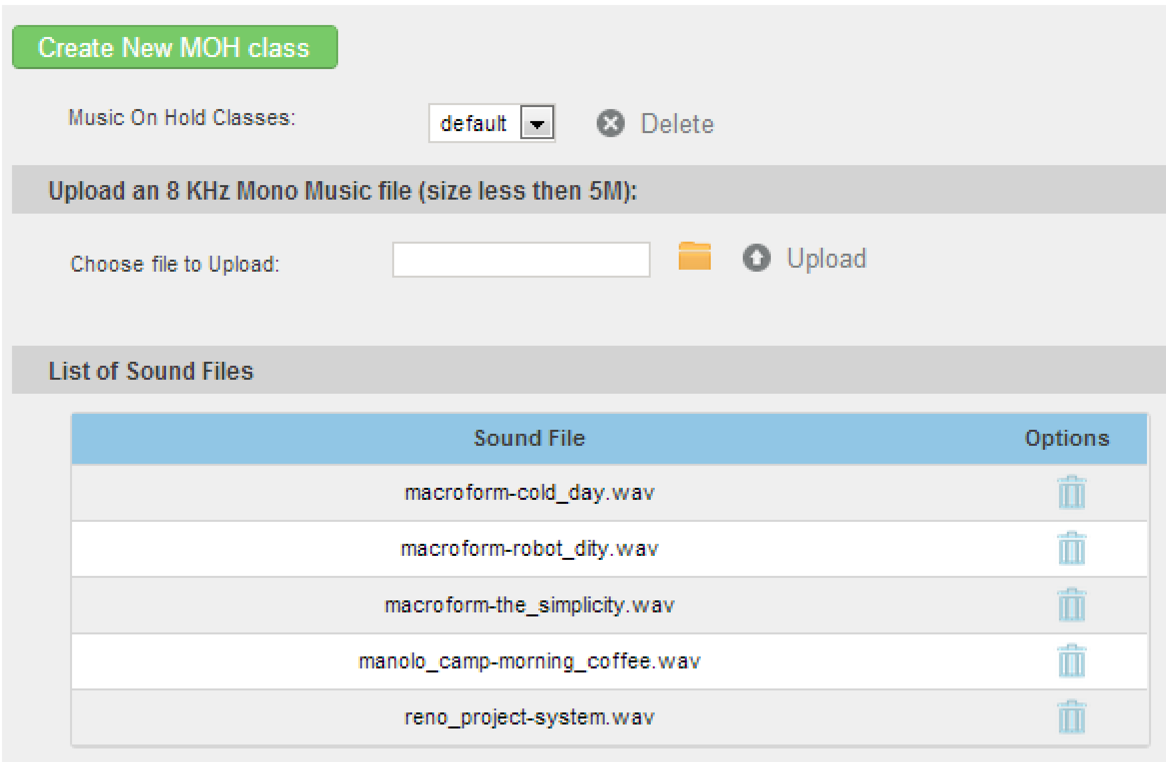


Figure 32: Music On Hold Default Class

- 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 less than 5M.
- Click on to delete the sound file for the selected Music On Hold file.

FAX/T.38

On the UCM6102/6104/6108/6116, the Fax extension can receive T.38 Fax to the specified Email address. Fax/T.38 settings can be accessed via Web GUI->**PBX**->**Internal Options**->**FAX/T.38**.

CONFIGURING 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 "Settings to Fax" to configure the following options.

Table 27: FAX/T.38 Settings

Enable Error Correction Mode (ECM)	Enable 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.

- Click on  to edit the Fax extension.

- Click on  to delete the Fax extension.

Note:
Users could also use Fax2Mail if "FaxDetect" option is enabled for the user and VoIP trunk. The Fax signal will be detected during talking, and the detected Fax will be received and sent to user's Email. If user's Email address is not configured, Fax will be sent to the default Email address in ***Table 27: FAX/T.38 Settings***. If the default Email address is empty, Fax will not be received.

INTERNAL OPTIONS

The configuration for PBX internal options can be accessed via Web GUI->**PBX**->**Internal Options**.

INTERNAL OPTIONS/GENERAL

General Preferences	
Global OutBound CID	This is the default global CallerID that is used for all outgoing calls when no other CallerID is defined. If ther "User" tab or "VoIP Trunks" tab does not have defined CallerID neither, this Global OutBound CID will be used for CallerID.
Global OutBound CID Name	This is the global CallerID Name that is used for all outgoing calls. If this value is defined, all outgoing calls will have a "CallerId Name" set to this value. Usually this value could be your company name. Leave this value blank if you would like to have the users' "CallerID Name" display on outbound calls.
Operator Extension	The operator extension is the number dialed when users press "0" to exit Voicemail. It's also available in IVR option.
Ring Timeout	Number of seconds to ring an extension before sending to the user's voicemail box.
Extension Preferences	
Enable Rand Password	If enabled, random password will be generated when the extension is created. The default setting is enabled. It is recommended to enable it for security purpose.
Disable Extension Range	If set to Yes, users could disable specified extension range. The default extension range assignment are: User Extension: 6000-6299 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 for PBX to work

	properly.

INTERNAL OPTIONS/FEATURE CODES

Call Feature	Description
Blind Transfer	• Default code: #1. • Enter the code during active call. After hearing "Transfer", enter the number to transfer to. Then the user will be disconnected. • Options - Neither: Disable the feature code. - Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Attended Transfer	• Default code: *2. • Enter the code during active call. After hearing "Transfer", 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 - Neither: Disable the feature code. - Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Disconnect	• Default code: *0. • Enter the code during active call to disconnect the call. • Options - Neither: Disable the feature code. - Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Call Parking	• Default code: #72. • Enter the code during active call to park the call. • Options - Neither: Disable the feature code. - Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Audio Record	• Default code: *1. • Enter the code to record the audio call. • Options - Neither: Disable the feature code.

	- Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Audio Mix Record	• Default code: *3. • Enter the code to record the audio call and the PBX will mix the streams natively on the fly as the call is in progress. • Options - Neither: Disable the feature code. - Caller Enable: Enable the feature code on caller side only. - Callee Enable: Enable the feature code on callee side only.
Do Not Disturb (DND) Active	• Default code: *77.
Do Not Disturb (DND) Deactive	• Default code: *78.
Call Forward Busy Active	• Default Code: *71. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Busy Deactive	• Default Code: *72.
Call Forward No Answer Active	• Default Code: *73. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward No Answer Deactive	• Default Code: *74.
Call Forward Uncondition Active	• Default Code: *75. • Enter the code and follow the voice prompt. Or enter the code followed by the extension to forward the call.
Call Forward Uncondiion Deactive	• Default Code: *76.
Feature Digit Timeout	• Default Setting: 1000. • This is the maximum timeout (in milliseconds) between the digits input of the feature code.
Extension to Dial to Park a Call	• Default Extension: 700. • During an active call, press to park the call.
Extensions for Parked Calls	• 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.
Parked Call Timeout (in secs)	• Default setting: 120. • This is the timeout for the call to be parked. After the timeout, if the call is not retrieved, the extension who parked the call will be called back.
Dial Voice Mail	• 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.
Voice Mail Main	• 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.
Call Pickup	• Default Code: **. • To pick up a call for extension xxxx, enter the code followed by the extension number xxxx.

INTERNAL OPTIONS/RTP SETTINGS

RTP Start	RTP port starting address. The default setting is 10000.
RTP End	RTP port ending address. The default setting is 20000.
Strict RTP	Enables/disables strict RTP protection. When enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
RTP Checksums	Enables/Disables RTP Checksums. The default setting is "Disable".

INTERNAL OPTIONS/HARDWARE CONFIG

Analog Hardware	
Options	Select Loop Start or Kewl Start for each FXS port and FXO port.
Tone Region	Select country for default tones (dial tone, busy tone, ring tone and etc.
Advanced Settings	
FXO Opermode	Specify 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. Select the country in the list. FCC is equivalent to United States. TBR21 is equivalent to Austria, Belgium, Denmark, Finland, France, Germany, Greece, Iceland, Ireland, Italy, Luxembourg, Netherlands, Norway, Portugal, Spain, Sweden, Switzerland, and the United Kingdom. If option is specified, FCC will be used by default.
ACIM Override	Check to override AC Impedance.
FXS Opermode	Specify 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. Select the country in the list. FCC is equivalent to United States. TBR21 is equivalent to Austria, Belgium, Denmark, Finland, France, Germany, Greece, Iceland, Ireland, Italy, Luxembourg, Netherlands, Norway, Portugal, Spain, Sweden, Switzerland, and the United Kingdom. The default setting is FCC.
TISS Override	Check to override Two-Wire Impedance Synthesis.
PCMA Override	Specifies the codec to be used for analog lines. North American users should choose PCMU. All other countries, unless otherwise known, should be assumed to be PCMA. If no user choice is specified, the default is PCMU.
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 during "Fast Ringer" operation. This option is used with "Fast Ringer" option. The default setting is "Normal".
Ring Detect	Configure ring detection. 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. • None: No detection • FSK: Frequency Shift Key detection • NEON: Neon MWI detection. The default setting is "None".

INTERNAL OPTIONS/STUN MONITOR

STUN Server	Configures the STUN server to query. Valid format: [(hostname | IP-address) [':' port] The default port number is 3478 if not specified. Leave this field blank to disable STUN.
STUN Refresh	Number of seconds between STUN Refreshes. The default setting is 30 seconds.

PBX SETTINGS

IAX SETTINGS

The UCM6102/UCM6104/UCM6108/UCM6116 IAX Settings can be accessed via Web GUI->**PBX**->**IAX Settings**.

IAX SETTINGS/GENERAL

Bind Port	Allows iax2 to listen to another port. The default setting is 4569.
Bind Address	Forces iax2 to bind to a specific address instead of all addresses. The default setting is 0.0.0.0.
IAX1 Compatibility	Enables/disables iax1 style compatibility.
No Checksums	Enables/disables checksums.
Delay Reject	Enables/disables iax2 to delay reject of calls to avoid DOS.
ADSI	Enables/disables ADSI phone compatibility.
Music On Hold Interpret	Specifies Music On Hold class.
Music On Hold Suggest	Suggests Music On Hold for the channel.
Language	Configures default language for the channel. This can be used by prompts.
Bandwidth	Configures allowed codecs for different bandwidth requirement. The default setting is "Low".

IAX SETTINGS/CODECS

The following codes are supported in UCM6102/UCM6104/UCM6108/UCM6116 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/JITTER BUFFER

Enable Jitter Buffer	Enables the use of jitter buffer on the receiving side of a SIP channel.
Force Jitter Buffer	Forces the use of jitter buffer on the receiving side of a SIP channel.
Drop Count	Configures drop count.
MAX Jitter Buffer	Configures the maximum time (in milliseconds) 0 for the buffer.
MAX Interpolation Frames	Configures the maximum number of interpolated frames the jitter buffer should return consecutively.
Recync Threshold	Jumps in the frame timestamps over where the jitter buffer is resynchronized. This feature is useful to improve the quality of voice with big jumps in/broken timestamps sent from exotic devices and programs. The default setting is 1000.
Max Excess Buffer	Configures the maximum number (in milliseconds) to pad the jitter buffer.
Min Excess Buffer	Configures the minimum number (in milliseconds) to pad the jitter buffer.
Jitter Shrink Rate	Configures the jitter shrink rate.

IAX SETTINGS/REGISTRATION

Min Reg Expire	Minimum duration (in seconds) of registrations/subscriptions. The default setting is 60.
Max Reg Expire	Maximum duration (in seconds) of incoming registration/subscriptions. The default setting is 3600.
IAX Thread Count	Configures number of IAX threads.
IAX Max Thread Count	Configures maximum number of IAX threads.
Auto Kill	When set to "yes", the connection will be terminated if ACK for the NEW message is not received in 2000ms. Users could also specify number (in milliseconds) in addition to "yes" and "no".
Authentication Debugging	Enables/disables IAX related debug output in log messages.
Codec Priority	Configures codec negotiation priority to Caller, Host, Disabled or Reqonly.
Type of Service	Configures ToS bit for preferred IP routing.
Trunk Frequency	Configures frequency of trunk frames measured in milliseconds.

Trunk Time Stamps	Enables/disables attaching time stamps to trunk frames.

IAX SETTINGS/SECURITY

Call Token Optional	A single IP address or a range of IP addresses for which call token validation is not required in the form 11.11.11.11 or 11.11.11.11/22.22.22.22.
Max Call Numbers	Limits the amount of call numbers allowed for a single IP address.
Max Nonvalidated Call Numbers	Limits the amount of nonvalidated call numbers for all IP addresses combined.
Call Number Limits	Limits the call numbers for a given IP range.

SIP SETTINGS

The UCM6102/UCM6104/UCM6108/UCM6116 IAX Settings can be accessed via Web GUI->**PBX**->**SIP Settings**.

SIP SETTINGS/GENERAL

Realm For Digest Authentication	Realms MUST be globally unique according to RFC 3261. Configure this value as your host or domain name. The default setting is \"asterisk\". If a system name is configured in asterisk.conf, this value will be set to the configured system name.
UDP Port to Bind to	The default setting is 5060.
IP Address to Bind to	The default setting is 0.0.0.0, which means binding to all addresses.
Domain	Use comma to separate a list of domains that the UCM6102/UCM6104/UCM6108/UCM6116 will be responsible for.
Allow Guest Calls	Enables/disables guest calls.
Overlap Dialing Support	Enables/disables dialing support.
Allow Transfer	Enables/disables all transfers (unless enabled in peers or users) initiated by the endpoint. The Dial() options 't' and 'T' are not related to whether SIP transfers are allowed or not.
Enable DNS SRV Lookups (on outbound calls)	Enables/disables DNS SRV lookups on calls.
MWI From	When sending MWI NOTIFY requests, this value will be used in the

	"From:" header as the \"name\" part. If no "fromuser" is configured, the \"user\" part of the URI in the "From:" header will be filled with this value as well.
From Domain	Configures the domain in the "From:" field of the SIP header. It may be required by some providers for authentication.
Auto Domain	When turned on, the UCM6102/UCM6104/UCM6108/UCM6116 will add local host name and local IP to domain list.
Allow External Domains	Allow requests for domains that are not served by the UCM6102/UCM6104/UCM6108/UCM6116.

SIP SETTINGS/CODECX

The following codes are supported in UCM6102/UCM6104/UCM6108/UCM6116 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

SIP SETTINGS/JITTER BUFFER

Enable Jitter Buffer	Enables/disables the use of jitter buffer on the receiving side of a SIP channel.
Force Jitter Buffer	Forces the use of jitter buffer on the receiving side of a SIP channel.
Log Frames	Enable/disables jitter buffer frame logging.
Max Jitter Buffer	Configures max length of the jitter buffer in milliseconds.
Resync Threshold	Jumps in the frame timestamps over where the jitter buffer is resynchronized. This feature is useful to improve the quality of voice

	with big jumps in/broken timestamps sent from exotic devices and programs. The default setting is 1000.
Implementation	The Jitter buffer implementation used on the receiving side of a SIP channel. Users could select "Fixed" (with size always equals to jbmaxsize) or "Adaptive" (with variable size which is the new jb of IAX2).

SIP SETTINGS/MISCELLANEOUS

Register	Register as a SIP user agent to a SIP proxy (provider).
Register Timeout	The interval (in seconds) for the UCM6102/UCM6104/UCM6108/UCM6116 to retry registration. The default setting is 20.
Register Attempts	Number of registration attempts before the UCM6102/UCM6104/UCM6108/UCM6116 gives up. The default setting is 0 (keep trying until the server side accepts the registration request).
Video Max Bitrate (kb/s)	Maximum bitrate (kb/s) for video calls. The default setting is 384.
Support for SIP Video	Enables/disables SIP video support.
Generate Manager Events	Generates manager events when SIP UA performs events (e.g. hold).
Reject NonMatching Invites	When rejecting an incoming INVITE or REGISTER request, always reject with "401 Unauthorized" instead of notifying the requester that if there is a matching user or peer for the request.
NonStandard G.726 Support	If the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs).

SIP SETTINGS/SESSTION TIMER

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

SIP SETTINGS/TLS AND TCP

TCP Enable	Enables/disables server for incoming TCP connections. The default setting is "No".
TCP Bindaddr	IP address for TCP server to bind to (0.0.0.0: binds to all interfaces). The default port number is 5060 if not specified.
TLS Enable	Enables/disables server for incoming TLS (secure) connections. The default setting is "No".
TLS Bindaddr	IP address for TLS server to bind to (0.0.0.0: binds to all interfaces). The default port number is 5061 if not specified. **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 Self Signed CA	This is the CA certificate is the TLS server being connected to requires self signed certificate, including server's public key. This file will be renamed as "asterisk.ca" automatically. **Note:** The size of your ca file can't be larger than 2MB.
TLS Cert	This is the Certificate file (*.pem format only) used for TLS connections. This file will be renamed as "asterisk.pem" automatically. **Note:** The size of your certificate can't be larger than 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 your certificate can't be larger than 2MB.
TLS CA List	The list of files under the CA Cert directory.

SIP SETTINGS/NAT

External Address	A static address (and port) that will be in outbound SIP messages if the

	UCM6102/UCM6104/UCM6108/UCM6116 is behind NAT. If it's a hostname, it will only be looked up only.
External Host	Specifies an external host, which is similar to External Address except the hostname will be looked up every "External Refresh" interval and Asterisk will perform DNS queries periodically.
External Refresh	Configures the refresh interval for the external host.
External TCP Port	Configures the externally mapped TCP port when the UCM6102/UCM6104/UCM6108/UCM6116 is behind a static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when UCM6102/UCM6104/UCM6108/UCM6116 is behind a static NAT or PAT. The default value is 5061.
Local Network Address	A list of network addresses that are considered inside of the NAT network. Multiple entries are allowed, e.g., a reasonable set could be as follows: 192.168.0.0/255.255.0.0
NAT Mode	This is a global NAT setting that will affects all peers and users. • No: Use rport if the remote side requires it. • Force rport: Force rport to always be on. This is the default setting. • Yes: Force rport to always be on and perform comedia RTP handling. • Comedia: Use rport if the remote side requires it and perform comedia RTP handling. **Note:** "comedia RTP handling" refers to the technique of sending RTP to the port where the other endpoint's RTP comes from. This can also be rephrased as "connection-oriented media".
Allow RTP Reinvite	When turned on, the UCM6102/UCM6104/UCM6108/UCM6116 will try to redirect the RTP media stream (audio) to go directly from the caller to the callee. • Yes: Enables RTP Reinvite. • NoNAT: Allows media path redirection (reinvite) but only when the peer is not be behind NAT. The RTP core can determine 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

The following options are provided to configure SIP ToS on the UCM6102/UCM6104/UCM6108/UCM6116.

ToS For Signaling Packets	Configure the Type of Service for SIP packets. The default setting is None.
ToS For RTP Audio Packets	Configure the Type of Service for RTP audio packets. The default setting is None.
ToS For RTP Video Packets	Configure the Type of Service for RTP video packets. The default setting is None.
Default Incoming/Outgoing Registration Time	Configure the default length of time (in seconds) of incoming/outgoing registration. The default setting is 120.
Max Registration/Subscription Time	Configure the maximum length of time (in seconds) of incoming registration and subscription allowed by the PBX. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum length of time (in seconds) of incoming registration and subscription allowed by the PBX. 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 in the dialplan and the peer channel placing the call on hold doesn't suggest the Music On Hold class neither. The default Music On Hold class will be used if not specified.
Music On Hold Suggest	Configure the Music On Hold class to suggest to the peer channel when placing the peer on hold. It can be specified globally or per user/per peer. The default Music On Hold class will be used if not specified.
Enable Relaxed DTMF	Configure to relax the DTMF handling. The default setting is disabled.
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.
RTP Timeout	If there is no RTP activity in the timeout (in seconds) configured in this option, the call will be terminated. The default setting is no timeout. This

	setting doesn't apply to calls on hold.
RTP Hold Timeout	When the call is on hold, if there is no RTP activity in the timeout (in seconds) configured in this option, 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 disabled.
Send Remote Party ID	Configure whether the Remote-Party-ID should be sent. The default setting is disabled.
Generate In-Band Ringing	Configure whether the PBX should generate inband ringing. If "Never" is selected, inband ringing will not be generated even the end point device is not working properly. The default setting is Never.
Server User Agent	Configure to replace the user agent string.
Allow Nonlocal Redirect	If enabled, 302 or Redirect is allowed to non-local SIP address. The default setting is disabled.
Add "user=phone" to URI	If enabled, "user=phone" will be added to URI that contains a valid phone number. The default setting is disabled.
Send Compact SIP Headers	If enabled, compact SIP headers will be sent. The default setting is disabled.
Time Between MWI Checks	Configure the default time (in seconds) between Mailbox checks for peers. The default setting is 10.
Min Roundtrip Time (T1 Time)	Configure the minimum roundtrip time (in milliseconds) of the messages sent to the monitored hosts. The default setting is 100.

SIP SETTINGS/DEBUG

Enable SIP Debugging	Enables/disables SIP debugging.
Record SIP History	Records SIP history.
Dump SIP History	Dumps SIP history at the end of SIP dialog.
Subscribe Context	Configures a specific context for SUBSCRIBE requests. This setting is useful to limit subscriptions to local extensions.
Allow Subscribe	Enables/disables support for subscriptions.
Notify on Ringing	Sends out NOTIFY on ringing status.

STATUS AND REPORTING

PBX STATUS

The UCM6102/UCM6104/UCM6108/UCM6116 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**.

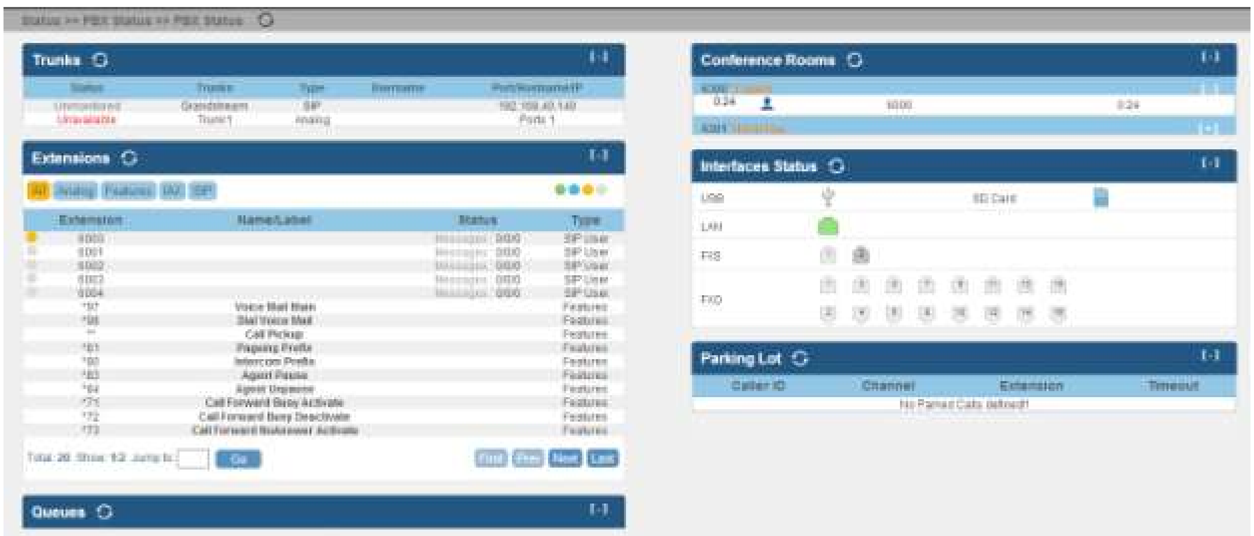


Figure 33: Status->PBX Status

TRUNKS

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

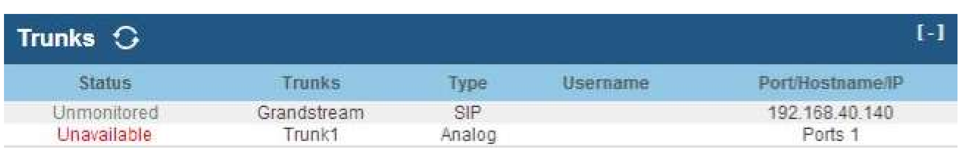


Figure 34: Trunk Status

Table 28: Trunk Status

Status	Displays trunk status. • For analog trunk, the possible status are: Available

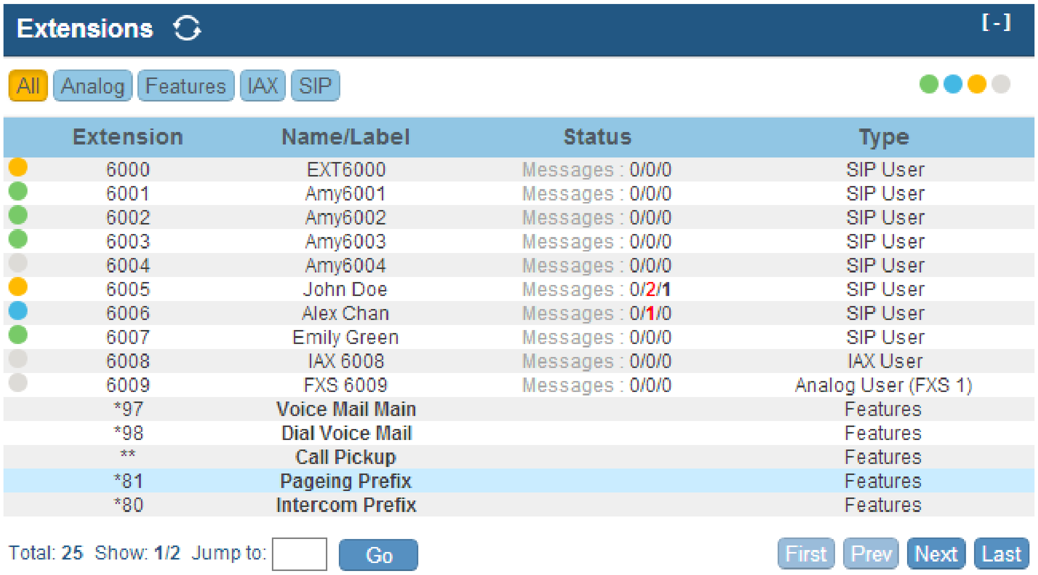
	Busy Unavailable Unknown Error • For SIP Peer trunk, the possible status are: Unreachable: The hostname cannot be reached. Unmonitored: QUALIFY feature is not turned on to be monitored. Reachable: The hostname can be reached. • For SIP Register trunk, the possible status are: Registered Unrecognized Trunk
Trunks	Displays trunk name
Type	Displays trunk Type. • Analog: This trunk is an analog trunk • SIP: This trunk is a SIP trunk • IAX: This trunk is an IAX trunk
Username	Displays username for this trunk.
Port/Hostname/IP	Displays 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.



Extension	Name/Label	Status	Type
6000	EXT6000	Messages : 0/0/0	SIP User
6001	Amy6001	Messages : 0/0/0	SIP User
6002	Amy6002	Messages : 0/0/0	SIP User
6003	Amy6003	Messages : 0/0/0	SIP User
6004	Amy6004	Messages : 0/0/0	SIP User
6005	John Doe	Messages : 0/2/1	SIP User
6006	Alex Chan	Messages : 0/1/0	SIP User
6007	Emily Green	Messages : 0/0/0	SIP User
6008	IAX 6008	Messages : 0/0/0	IAX User
6009	FXS 6009	Messages : 0/0/0	Analog User (FXS 1)
*97	Voice Mail Main		Features
*98	Dial Voice Mail		Features
**	Call Pickup		Features
*81	Pageing Prefix		Features
*80	Intercom Prefix		Features

Figure 35: Extension Status

Table 29: Extension Status

Extension	Displays extension number (including feature code). The color indicator has the following definitions. • Green: Free • Blue: Ringing • Yellow: In Use • Grey: Unavailable
Name/Label	Displays name (callerID name) or label (feature code function) for the extension.
Status	Displays 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 read already.
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.

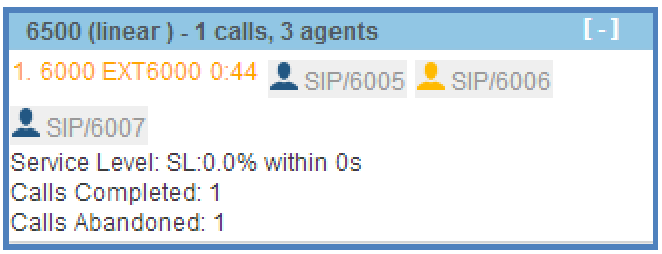


Figure 36: Queue Status

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 30: Agent Status

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

On the UCM6102/UCM6104/UCM6108/UCM6116, **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 and call duration for each user, as well as conference duration.

Figure 37: 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 UCM6102/UCM6104/UCM6108/UCM6116. The following example shows the interface status for UCM6116 with USB, SD card, LAN port, FXS1 connected.

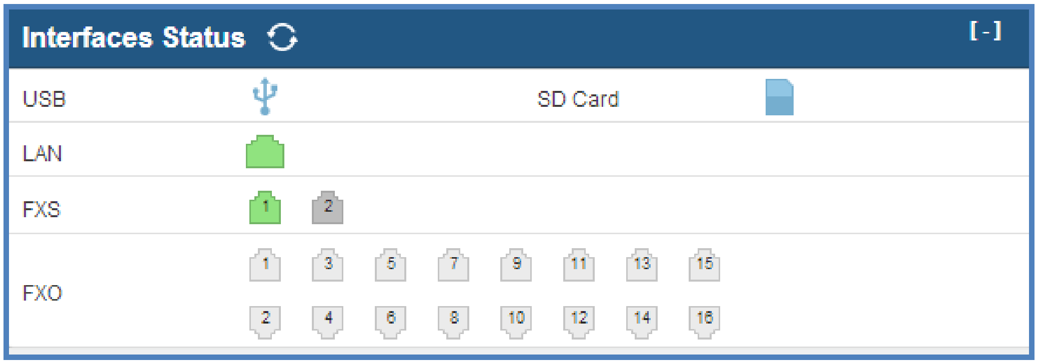


Figure 38: UCM6116 Interfaces Status

Table 31: Interface Status Indicators

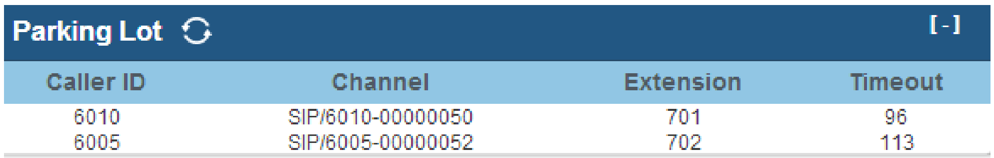
	USB connected.
	USB disconnected.
	SD Card connected.
	SD Card disconnected.
	LAN/WAN connected.
	LAN/WAN not configured.
	LAN/WAN disconnected.
	FXS/FXO connected.
	FXS/FXO waiting.
	FXS/FXO busy.
	FXS/FXO not configured.
	FXS/FXO disconnected.

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 UCM6102/UCM6104/UCM6108/UCM6116 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 39: Parking Lot Status

Table 32: Parking Lot Status

Caller ID	Displays the caller ID who parks the call.
Channel	Displays channel for the call park.
Extension	Displays the parking lot number where the call is parked/retrieved.
Timeout	Displays 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 parked 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 UCM6102/UCM6104/UCM6108/UCM6116 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 UCM6102/UCM6104/UCM6108/UCM6116. Please see details in the following table.

Table 33: 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 UCM6102/UCM6104/UCM6108/UCM6116. Please see details in the following table.

Table 34: 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
  Asterisk server 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.

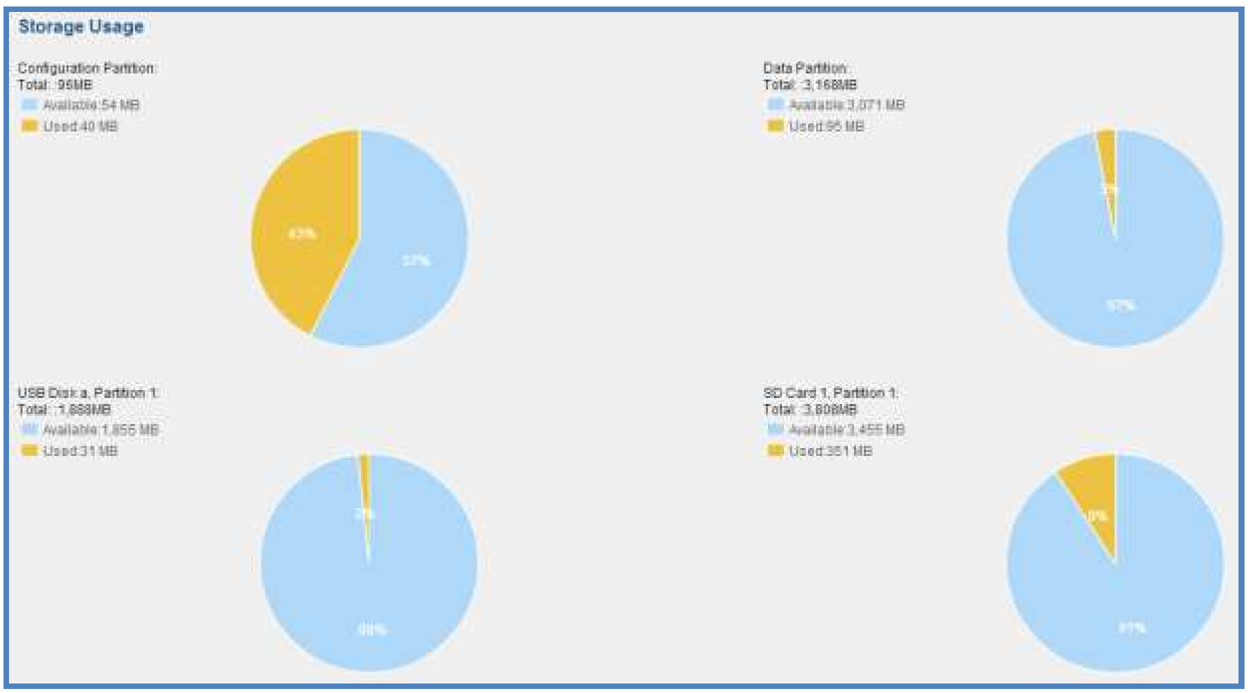


Figure 40: System Status->Storage Usage

RESOURCE USAGE

When configuring and managing the UCM6102/UCM6104/UCM6108/UCM6116, 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.

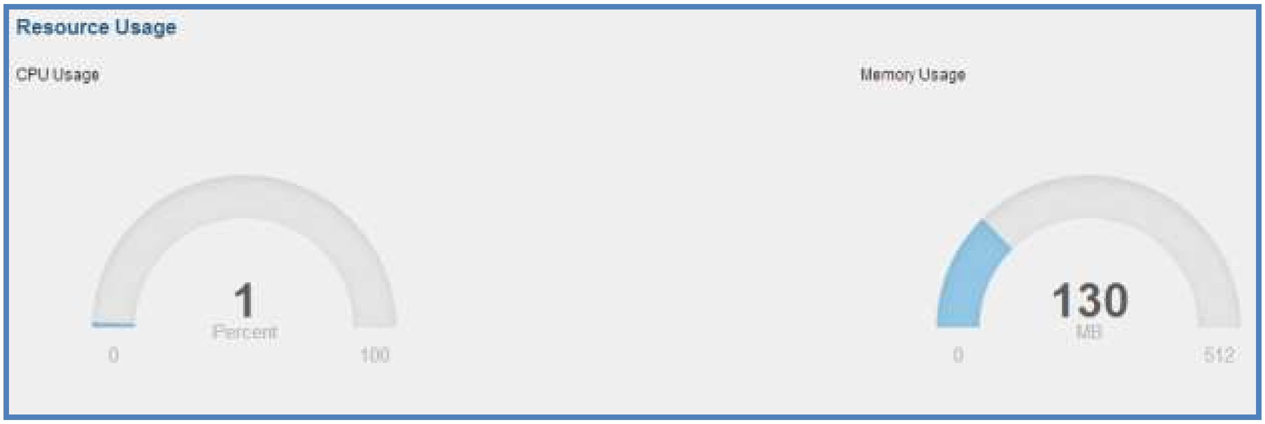


Figure 41: System Status->Resource Usage

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 UCM6102/UCM6104/UCM6108/UCM6116.

- **Start Time**. Format: 2013-03-27 16:47:03.
- **Duration**. Format: 0:00:10.
- **Source**. Format: 6012.
- **Destination**. Format: 6005.
- **Caller ID**. Format: "John Doe"<6012>.
- **Disposition**. Format: NO ANSWER, BUSY, ANSWERED, or FAILED.

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 the "View Report" button to display the generated report.

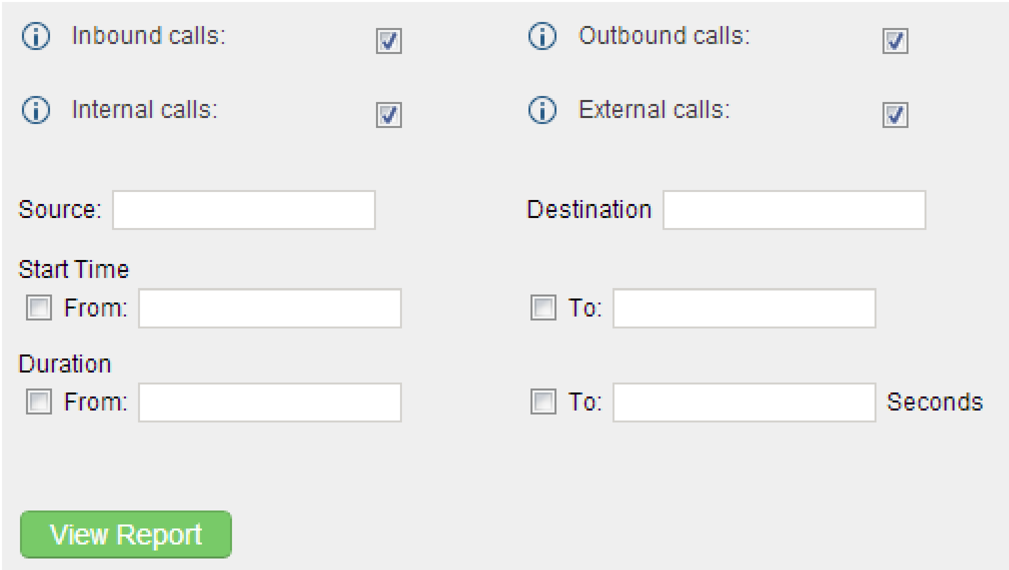


Figure 42: CDR Filter

Table 35: 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.

The call report will display as the following figure shows.

No.	Start Time	Duration	Source	Destination	Caller ID	Disposition	Options
1	2013-03-08 01:37:03	0:00:02	1003	1002	"1003-214" <1003>	NO ANSWER	
2	2013-03-08 01:37:11	0:00:03	2022	2003	"2022-228" <2022>	FAILED	
3	2013-03-08 01:37:40	0:00:03	2021	1002	"2021-228" <2021>	ANSWERED	
4	2013-03-08 01:41:59	0:00:03	2023	81003	"2023-228" <2023>	FAILED	
5	2013-03-08 01:42:16	0:00:02	2023	81002	"2023-228" <2023>	FAILED	
6	2013-03-08 01:42:40	0:00:03	2023	1002	"2023-228" <2023>	ANSWERED	
7	2013-03-08 01:49:24	0:00:07	1002	2023	"1002-214" <1002>	ANSWERED	
8	2013-03-08 01:49:43	0:00:06	1003	2022	"1003-214" <1003>	ANSWERED	
9	2013-03-11 17:14:38	0:00:04	6002	7000	"6002" <6002>	FAILED	
10	2013-03-11 17:15:09	0:00:38	6002	6004	"6002" <6002>	ANSWERED	

Figure 43: 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" one time to sort the report according to start time. Clicking on "Start Time" again to 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 first row has audio recording file for the call.

1	2013-03-27 20:25:31	0:00:12	6000	6010	"EXT6000" <6000>	ANSWERED	
2	2013-03-27 20:25:18	0:00:08	6000	6010	"EXT6000" <6000>	ANSWERED	

Figure 44: Call Report Entry With Audio Recording File

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).

CDR Statistics is an additional feature on the UCM6102/UCM6104/UCM6108/UCM6116 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.

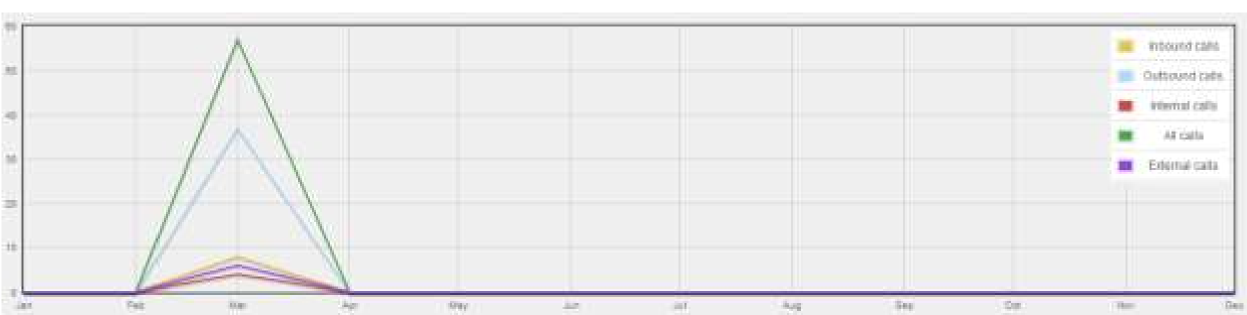


Figure 45: CDR Statistics

Table 36: 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

UPGRADING VIA NETWORK

The UCM6102/UCM6104/UCM6108/UCM6116 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**.

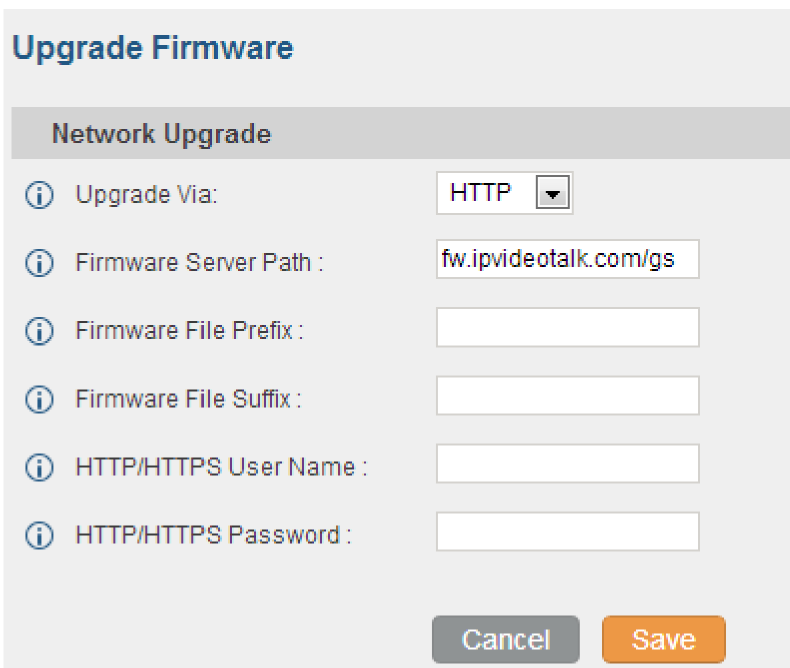


Figure 46: Network Upgrade

Table 37: 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 UCM6102/UCM6104/UCM6108/ UCM6116.
Firmware File Suffix	If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the UCM6102/UCM6104/UCM6108/ UCM6116.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.

Click on "Save" and "Apply Changes". Then reboot the device to start the upgrading process.

UPGRADING VIA LOCAL UPLOAD

If there is no HTTP/TFTP server, users could also upload the firmware to the UCM6102/UCM6104/UCM6108/UCM6116 directly via Web GUI. Please follow the steps below to upload firmware locally.

- Download the latest UCM6102/UCM6104/UCM6108/UCM6116 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.
- Click on to start upgrading.

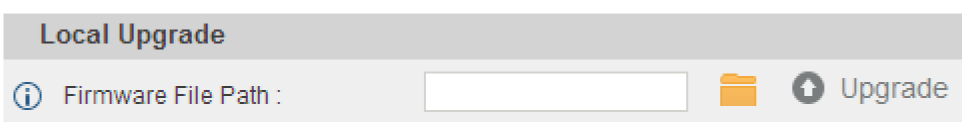


Figure 47: Local Upgrade

- Wait until the upgrading process is successful and a window will be popped up. Click on "OK" to reboot the UCM6102/UCM6104/UCM6108/UCM6116 and check the firmware version when it boots up.

⚠ **Note:**

Please do not interrupt or power cycle the UCM6102/UCM6104/UCM6108/UCM6116 when the upgrading process is on.

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://support.solarwinds.net/updates/New-customerFree.cfm
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 UCM6102/UCM6104/UCM6108/UCM6116 device 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 UCM6102/UCM6104/UCM6108/UCM6116's web configuration interface;
5. Configure the Firmware Server Path to the IP address of the PC;
6. Update the changes and reboot the UCM6102/UCM6104/UCM6108/UCM6116.

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

BACKUP

LOCAL BACKUP

Users could backup the 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 device.
- 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.

Once backup is done, the list of the backups will be displayed with date and time. Users then can download, restore or delete it from the UCM6102/UCM6104/UCM6108/UCM6116 or the external device.

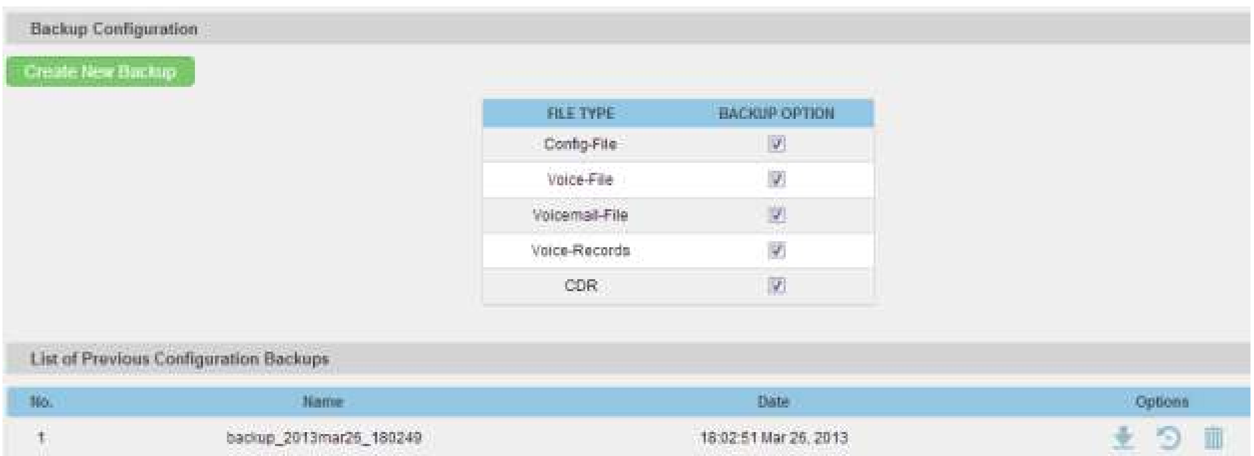


Figure 48: Local Backup

NETWORK BACKUP

Users could backup the voice records/voice mails/CDR/FAX in a daily basis via SFTP protocol automatically under Web GUI->**Maintenance**->**Backup**->**Network Backup**.

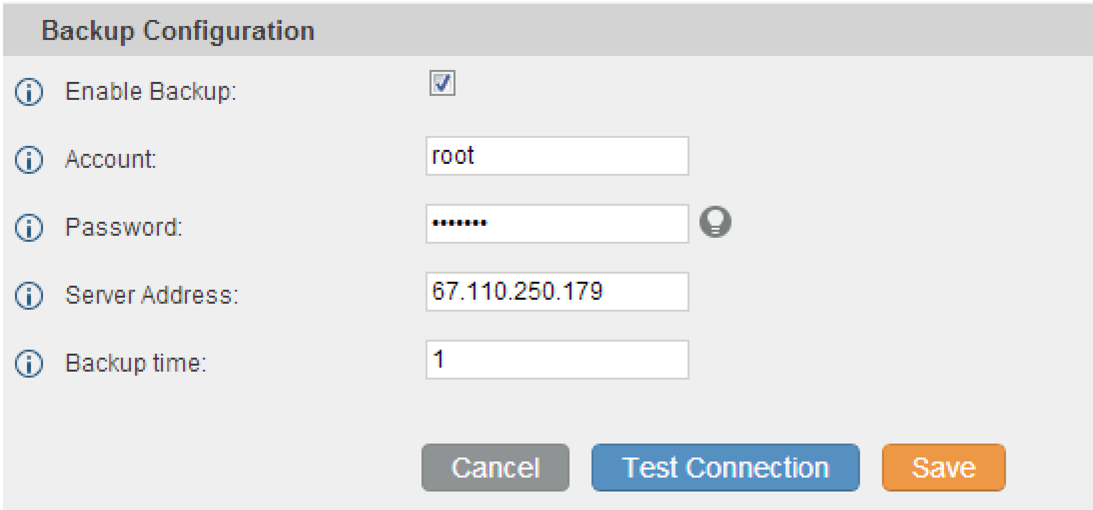


Figure 49: Network Backup

Table 38: Network Backup Configuration

Enable Backup	Enable the auto backup function.
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.

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

CLEANER

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

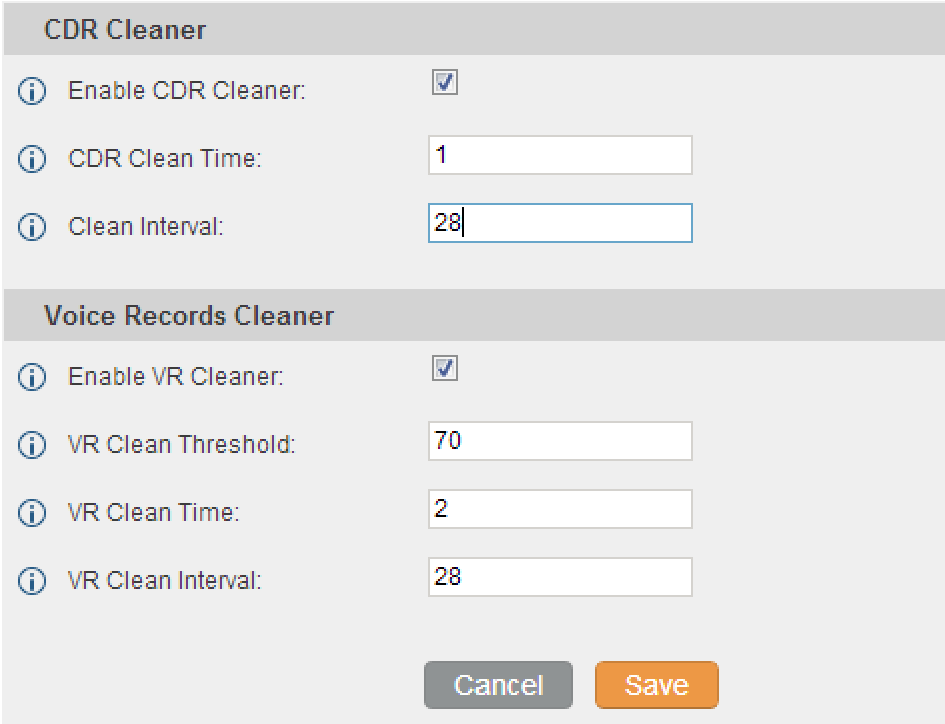


Figure 50: Cleaner

Table 39: 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 Configuration: All the Extensions, Trunks and Routing configurations, as well as the local settings (network settings, upgrading setting and etc) will be cleared.
- 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.

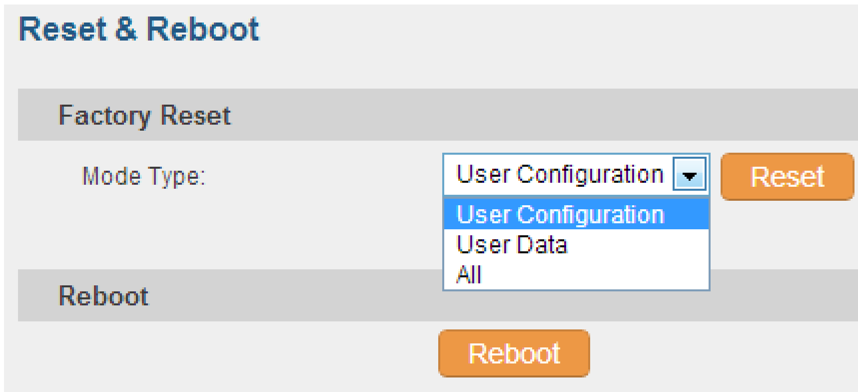


Figure 51: Reset and Reboot

SYSLOG

On the UCM6102/UCM6104/UCM6108/UCM6116, users could dump to 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.

TROUBLESHOOTING

On the UCM6102/UCM6104/UCM6108/UCM6116, 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.

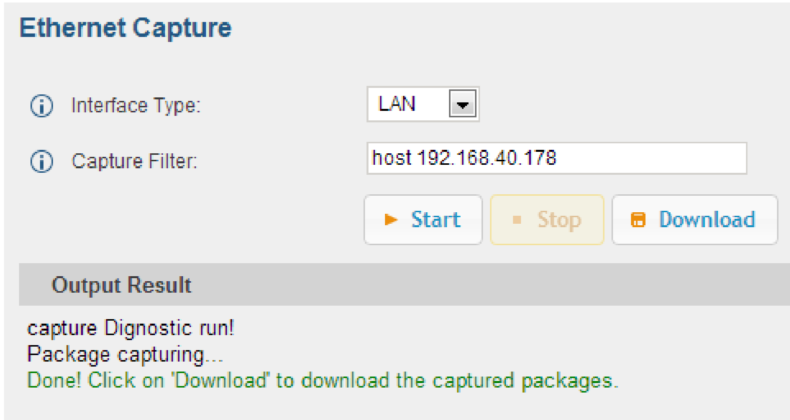


Figure 52: Ethernet Capture

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.

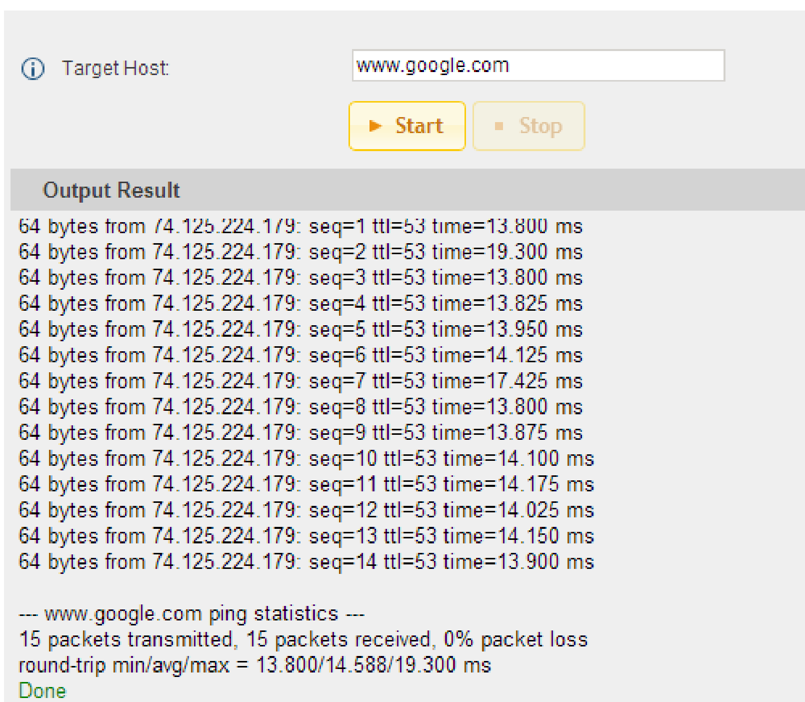


Figure 53: PING

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.

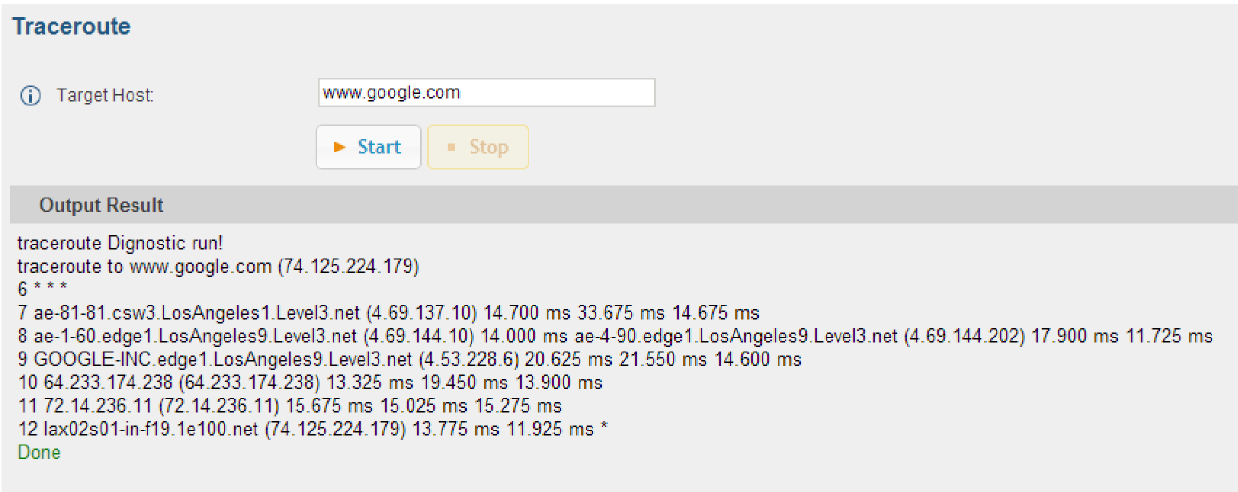


Figure 54: Traceroute

EXPERIENCING THE UCM6102/UCM6104/UCM6108/UCM6116

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/UCM6104/UCM6108/UCM6116, it will be sure to bring convenience and color to both your business and personal life.

Compliance

FCC Notice

This device complies with part15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

—Reorient or relocate the receiving antenna.
—Increase the separation between the equipment and receiver.
—Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
—Consult the dealer or an experienced radio/TV technician for help.

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Regulatory Information
U.S. FCC Part 68 Statement
This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. The unit bears a label on the back which contains among other information a product identifier in the format US: GNIIS00BUCM6104. If requested, this number must be provided to the telephone company.

This equipment uses the following standard jack types for network connection: RJ11C.

This equipment contains an FCC compliant modular jack. It is designed to be connected to the telephone network or premises wiring using compatible modular plugs and cabling which comply with the requirements of FCC Part 68 rules.

The Ringer Equivalence Number, or REN, is used to determine the number of devices which may be connected to the telephone line. An excessive REN may cause the equipment to not ring in response to an incoming call. In most areas, the sum of the RENs of all equipment on a line should not exceed five (5.0).

In the unlikely event that this equipment causes harm to the telephone network, the telephone company can temporarily disconnect your service. The telephone company will try to warn you in advance of any such disconnection, but if advance notice isn't practical, it may disconnect the service first and notify you as soon as possible afterwards. In the event such a disconnection is deemed necessary, you will be advised of your right to file a complaint with the FCC.

From time to time, the telephone company may make changes in its facilities, equipment, or operations which could affect the operation of this equipment. If this occurs, the telephone company is required to provide you with advance notice so you can make the modifications necessary to obtain uninterrupted service.

There are no user serviceable components within this equipment. See Warranty flyer for repair or warranty information.

It shall be unlawful for any person within the United States to use a computer or other electronic device to send any message via a telephone facsimile unless such message clearly contains, in a margin at the top or bottom of each transmitted page or on the first page of the transmission, the date and time it is sent and an identification of the business, other entity, or individual sending the message and the telephone number of the sending machine or of such business, other entity, or individual. The telephone number provided may not be a 900 number or any other number for which charges exceed local or long distance transmission charges. Telephone facsimile machines manufactured on and after December 20, 1992, must clearly mark such identifying information on each transmitted message. Facsimile modem boards manufactured on and after December 13, 1995, must comply with the requirements of this section.

This equipment cannot be used on public coin phone service provided by the telephone company. Connection to Party Line Service is subject to state tariffs. Contact your state public utility commission, public service commission, or corporation commission for more information.