Grandstream Networks, Inc.

GXW42XX series

16, 24, 32 or 48 FXS Ports

Analog IP Gateway

User Manual

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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 guide, could void your manufacturer warranty.

Safety Compliance

The GXW42XX adaptor complies with FCC/CE and various safety standards. The GXW42XX power adaptor is compliant with UL standard. Only use the universal power adapter provided with the GXW42XX package. The manufacturer's warranty does not cover damages to the device caused by unsupported power adaptors

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CHANGE LOG

This section documents significant changes from previous versions of GXW42XX 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 1.0.5.43

- Added feature Conference Party Hangup Tone. [Conference Party hangup tone]
- Added feature Inter-port Calling. [Inter-port Calling]
- Added feature LED Pattern. [LED Pattern]
- Added feature Reply to ICMP. [Reply to ICMP]
- Added feature Validate Server Certificates. [Validate Server Certificates]
- Added feature Load CA Certificates. [Load CA Certificates]
- Added feature Disable User Level Web Access. [Disable User Level Web Access]
- Added feature Disable Viewer Level Web Access. [Disable Viewer Level Web Access]
- Added feature Randomized Automatic Upgrade. [Randomized Automatic Upgrade]
- Added feature Virtual Network Interface. [Network settings for Management Interface]
- Added feature Disable Voicemail Reminder Tone. [Disable Voicemail Reminder Tone]
- Added feature Disable # as Redial Key. [Disable # as Redial key]
- Added feature Ringing Frequency. [Ringing Frequency]
- Added support for call waiting tone to be repeated while the caller is still calling. [Call Waiting]
- Added support to play stutter tone when call feature under Profiles → Call Features Settings is enabled. [Enable Call Features]

Firmware version 1.0.5.36

- Added support for Allow Auto-Dial Config Per Port. [Allow Auto-Dial Config Per Port]
- Added RTCP support. [RTCP]
- Added Custom Certificate support. [Custom Certificate]
- Added support for HTTPS server based on TLSv1.2.

Firmware version 1.0.5.30

- No major changes.

Firmware version 1.0.5.28

- Added option to specify the priority for caller ID display [Caller ID Display]
- Added option to save syslog locally. [Save Syslog]

- Added option “Prefer Primary Outbound Proxy” to enable registration through primary outbound proxy if registration expires. [Prefer Primary Outbound Proxy]
- Added feature “White list for WAN side” and “Black list for WAN side” for remote management. [Access Control Lists]
- Added feature” Lockout time interval” to change banning time after too many failed login attempts. [Lockout Time Interval].
- Added option “Web Access Mode” to choose between “HTTPS” and “HTTP” to access device Web UI. [Web Access Mode]
- Added option “HTTPS Web Port” to set HTTPS web port instead of using default web port. [HTTPS Web Port]
- Added local firmware upgrade support. [Upload Device Firmware]
- Change “Disable Telnet” option to “Disable SSH” option for more security. [Disable SSH]

Firmware version 1.0.5.16

- Added option [DTMF-RELAY Tag Respect SIP INFO] [Use Request Routing ID in SIP Headers] for Profile 1-4 audio settings.
- Added the option of [Crypto Life Time] to support RTP settings.
- Added option [Play busy/reorder tone before Loop Current Disconnect] for Profile 1-4 audio settings.
- Added the option of [Disable Connected Line ID] to support not display the connected line ID.
- Added the option of [Flash Digit Control] [Crypto Life Time] to overrides the default settings for call control.
- Added option to configure [Local RTP port].
- Added the option to configure [SIP Registration Failure Retry Wait Time upon 403 Forbidden].
- Added the option to enable/disable [Hold Target Before Refer].
- Added the option to enable/disable [Authenticate server certificate domain].
- Added the option to enable/disable [Authenticate server certificate chain].
- Added the option to configure [Trusted CA]
- Added the function of [Automatic Reboot].
- Added the option to [Allow DHCP Option 42 to NTP Server] to support Time and Date settings.
- Added the option to [Allow DHCP Option 2 to override time zone] to support Time and Date settings.
- Added the option to enable/disable [3CX Auto Provision] to support multicast SUBSCRIBE for provision.

Firmware version 1.0.5.8

- Added option [SIP REGISTER Contact Header Uses] [Use Request Routing ID in SIP Headers] for Profile 1-4 to support NAT transfer feature.

Firmware version 1.0.5.5

- Added option [Enable Fast RFC2833] to set DTMF duration by RFC2833.
- Added SNMP support. Added [SNMP] section.
- Added QoS settings for SIP and RTP packets. Rearranged options [Layer 2 QoS Settings] and [Layer 3 QoS Settings].
- Added star code [*98] to play registration ID
- Added star code [*99] to trigger provisioning process.
- Added support to download device configuration file.

Firmware version 1.0.4.22

- Added support for audio codec G.722
- Added option [Always send HTTP Basic Authentication Information] to enable sending HTTP authentication without server challenge
- Added the options to enable/disable [Use P-Preferred-Identity Header] and [Use Privacy Header]
- Added the options to enable/disable [Do Not Escape '#' as %23 in SIP URI]

Firmware version 1.0.4.17

- Added [Use Request Routing ID in SIP Headers] to enable/disable device to use configured Request URI Routing ID for certain FXS port in its SIP message Header when a trunk SIP account is used.
- Added option [Add Auth Header on Initial REGISTER] to enable/disable including authentication header in the initial REGISTER.

Firmware version 1.0.4.7

- Added option [Enable LLDP] to enable/disable LLDP.
- Added field [Request URI Routing ID] to allow device to route the calls to individual FXS ports based on the DID, when only have one sip registration.

Firmware version 1.0.4.4

- Added option [Display Language] to choose language for LCD display.
- Added option [Prompt Dial Tone Code] to configure dial code.

WELCOME

Thank you for purchasing the Grandstream GXW42XX Analog FXSIP Gateway. The GXW42XX offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GXW42XX supports popular voice codecs and is designed for full SIP compatibility and interoperability with third party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GXW FXS Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, and direct IP-IP Calling. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

Gateway GXW42XX Overview

The new GXW42XX series has a compact and quiet design and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The GXW42XX series features 16, 24, 32 or 48port FXS interface for analog telephones, dual 10/100/1000Mbps network ports, and RJ21analog port. In addition, it supports the option of 4 SIP Server profiles, caller ID for various countries/regions, T.38 fax, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codec including G.711 (a/u-law), G.723.1, G.726(16/24/32/40 bit rates), iLBC and G.729.

Safety Compliance

The GXW42XX is compliant with various safety standards including FCC/CE. Its power adapter is compliant with UL standard.

Warning: *Use only the power adapter included in the GXW42XX package. Use of alternative power adapter may permanently damage the unit.*

Warranty

Grandstream has a reseller agreement with our reseller customers. End users should contact the company from whom the product was purchased, for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Support for an RMA (Return Materials Authorization) number. Grandstream reserves the right to change the warranty policy without prior notification.

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.*

This document contains links to GXW42XX GUI Interfaces. Please download these examples from http://www.grandstream.com/sites/default/files/Resources/gxw42xx_gui.zip for your reference.

This document is subject to change without notice. The latest electronic version of this user manual is available for download at:
http://www.grandstream.com/sites/default/files/Resources/gxw42xx_usermanual_english.pdf

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CONNECT YOUR GXW42XX GATEWAY

Connecting the GXW42XX gateway is easy. Before you begin, please verify the contents of the GXW42XX package.

Equipment Packaging

Unpack and check all accessories. Equipment includes:

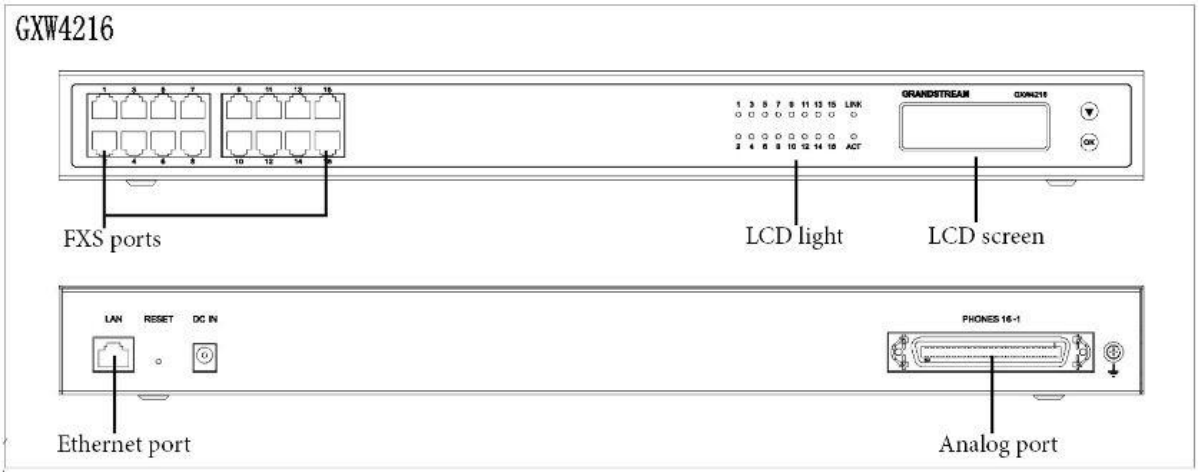
- one device unit
- one RJ45 Ethernet cable
- one 12V 5A universal power adapter (24V 6.25A for GXW4248)
- mount

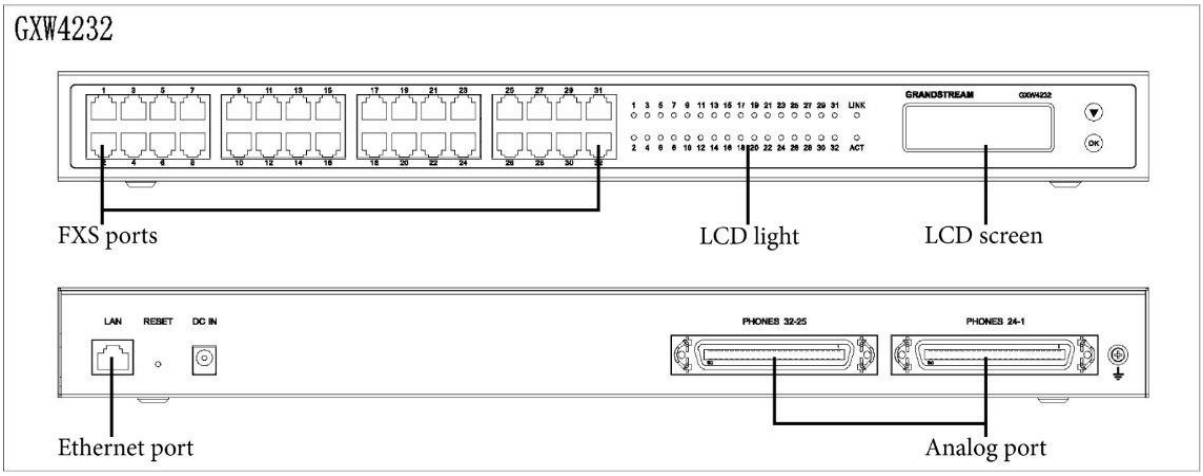
Connect the GXW42XX

Follow these four (4) steps to connect your GXW42XX gateway to the Internet and access the unit's configuration pages.

1. Connect standard touch-tone analog phones to the GXW42XX's RJ21 port with a RJ11 to R21 cable.
2. Insert anRJ45 Ethernet cable into the WAN port of GXW42XX and connect the other end to an uplink port (a router or a modem, etc.)
3. Plug the power adapter into the GXW42XX gateway into a power outlet.

Follow the instructions from the topic "Configuring GXW 42XX with Web Browser" for initial configuration. The GUI pages will guide you through the remaining steps to set-up your gateway.





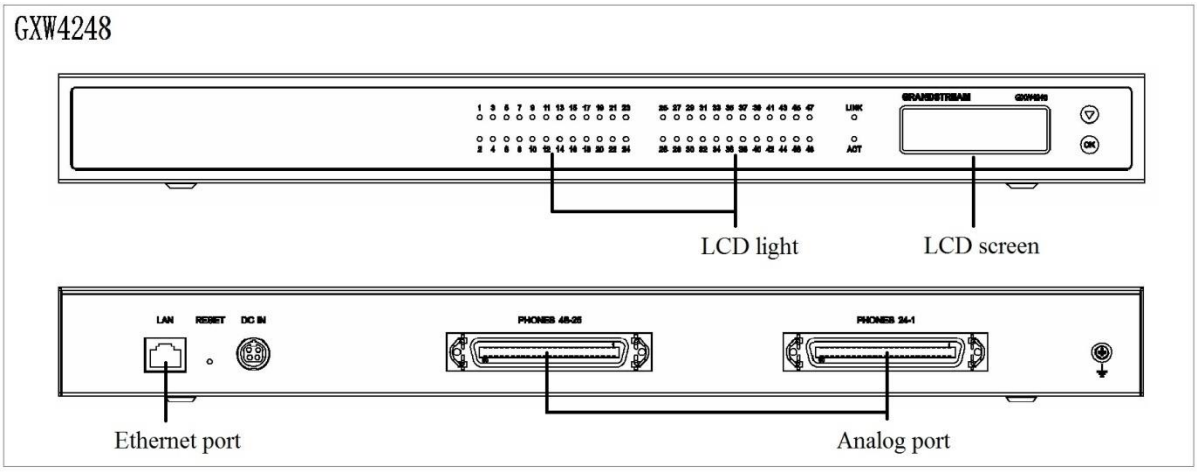


Figure 1: Diagram of GXW4216/24/32/48 Panel

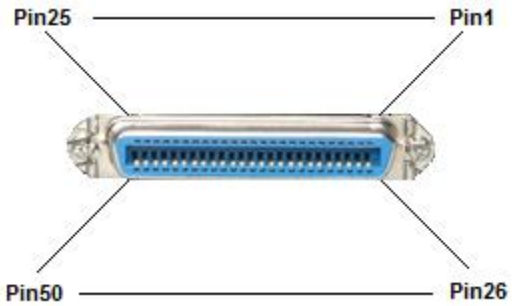
Table 1: Definitions of the GXW Connectors

Ethernet port	Connect to the internal LAN network or router
RESET	Factory Reset button. Press and hold for a while to reset factory default settings.
DC IN	Power adapter connection
Analog port	Connect to analog phones / fax machines with an RJ21 to RJ11 cable
FXS ports	FXS port to be connected to analog phones / fax machines.

Note: Once the GXW42XX is turned on and configured, the front display panel indicates the status of the unit.

Table 2: Definitions of the GXW Display Panel

Act LED	Remains ON if plug the network cable.
LINK LED	Indicates Ethernet port activity.
FXS LED	Indicate status of the respective FXS Ports on the back panel **Busy** - ON (Solid Green) **Available** - OFF Slow blinking FXS LEDs indicates Voice Mail for that port. All FXS LEDs slow blinking indicates provisioning.



Pin	Signal	Pin	Signal
1	RING1	26	TIP1
2	RING2	27	TIP2
3	RING3	28	TIP3
4	RING4	29	TIP4
5	RING5	30	TIP5
6	RING6	31	TIP6
7	RING7	32	TIP7
8	RING8	33	TIP8
9	RING9	34	TIP9
10	RING10	35	TIP10
11	RING11	36	TIP11
12	RING12	37	TIP12
13	RING13	38	TIP13
14	RING14	39	TIP14
15	RING15	40	TIP15
16	RING16	41	TIP16
17	RING17	42	TIP17
18	RING18	43	TIP18
19	RING19	44	TIP19
20	RING20	45	TIP20
21	RING21	46	TIP21
22	RING22	47	TIP22
23	RING23	48	TIP23
24	RING24	49	TIP24

Figure 2: 50 Pin Telco Connector

Note:

- For GXW4216, only first 16 pairs are connected.
- For GXW4232, first 24 pairs on 1st RJ21 connector and first 8 pairs on the 2nd RJ21 connector are connected.
- For GXW4224 and GXW4248, the last pair of pin 25 and 50 on each RJ21 is not connected.
- All LED lights display green when ON.
- LINK LED and ACT LED blinking in following status indicate different network speed.

10M: LINK and ACT all blink, ACT blinks faster than LINK

100M: LINK and ACT all blink, ACT blinks the same fast as LINK

1000M: LINK does not blink, only ACT blink

- Slow blinking of FXS LED together indicates a firmware upgrade or provisioning state.

GXW42XX FEATURES

The GXW42XX is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, Soft switches and SIP platforms. The GXW42XX FXS gateway is auto-configurable, remotely manageable and scalable. The GXW42XX gateways come in four models - the GXW4216, GXW4224, GXW4232and GXW4248, each offering superb voice quality, traditional telephony functionality, easy deployment, and 16, 24,32 and 48FXS ports respectively. Each model features flexible dialing plans, integrated call routing to support a pure IP network call and an external power supply.

Software Features Overview

- 16, 24, 32 or 48FXS ports (no front panel FXS ports on GXW4248)
- RJ-45 Ethernet ports
- 4 configurable SIP profiles
- Supports Voice Codecs:

  G711 (a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B, iLBC, T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation,
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets or analog trunks

TABLE 3: GXW42XX SOFTWARE FEATURES

	GXW4216	GXW4224	GXW4232	GXW4248
Telephone Interfaces	16 FXS ports	24 FXS ports	32 FXS ports	48 FXS ports
SIP Provisioning	16 SIP accounts, 4 profiles	24 SIP accounts, 4 profiles	32 SIP accounts, 4 profiles	48 SIP accounts, 4 profiles
Network Interface	10/100/1000 Mbps, RJ-45			

Number of Concurrent Calls (except when using SRTP)	16 Concurrent Calls	24 Concurrent Calls	32 Concurrent Calls	48 Concurrent Calls
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), LEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711			
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.722, G.723.1A, G.726(ADPCM with 16/24/32/40-bit rates), G.729, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header			
DHCP Server/Client	DHCP Client only			
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.21, V.27ter, V.29 for T.38 fax relay			
QoS	DiffServ, TOS, 802.1P/Q VLAN tagging			
Transport Protocol	RTP			
DTMF Method	In-audio, RFC2833, and/or SIP Info			
IP Signaling	SIP (RFC 3261)			
Provisioning	TFTP, HTTP, HTTPS			
Security	SRTP, TLS/SIPS, HTTPS			
Management	Syslog support, HTTP/HTTPS and SSH access			
Dial Plan	Yes			
3-Way Conference	3-Way conference with local mixing			
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, FSK and DTMF-based CID			
Polarity Reversal / Wink	Yes			
Network Connectivity	IPv4, TCP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SSH, PPPoE, STUN			

Hardware Specification

The hardware specifications of the GXW FXS series are detailed in Table 4.

Table 4: Hardware Specification of GXW42XX Series Gateways

	GXW4216	GXW4224	GXW4232	GXW4248
Telephone Interface	16 RJ11 Ports/ 1 RJ21 Port	24 RJ11 Ports/ 1 RJ21 Port	32 RJ11 Ports/ 2 RJ21 Ports	2 RJ21 Ports only
FXS LEDs	16	24	32	48
Network interface	LAN, Single 10/100/1000 BASE-TX, RJ45			
Power Input	**Input**:100-240VAC, 50/60Hz **Output**: 12V DC, 5.0A (4216/24/32) 24V DC, 6.25A (4248 only)			
LCD screen	128×32 pixel			
Telco connector	1 RJ21 (50 pins)	1 RJ21 (50 pins)	2 RJ21 (50 pins)	2 RJ21 (50 pins)
RJ-11 connectors	Yes	Yes	Yes	No
Function Buttons	1 button for Reset/Factory Reset			
Environmental	Operation: 0°C to 45°C Storage: -20°C to 60°C Humidity: 10% to 90% Non-condensing			
Mounting	Desktop and Rack mount			
Onhook Voltage	Fixed, 48V			
Ring Voltage	50Vrms (balanced ringing)			
Ring Frequency	20-50Hz			
Short Haul Loop	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire
Outdoor Protection	Over-voltage Protection and surge immunity			
Signaling	FXS Loop-start			

EMC	EN55032/EN55035 and FCC part15 Class B
Safety	UL
Compliance	FCC, CE, C-Tick

GXW42XX LCD Menu

The GXW42XX gateway series includes a small LCD screen for the display of basic information. The LCD has a display area of 128x32 pixels, which will allow for 2 lines of text with a 16px height limitation per line.

The LCD menu is showed as Figure 3: LCD Menu. Menu is navigated by the Down-arrow and OK button.

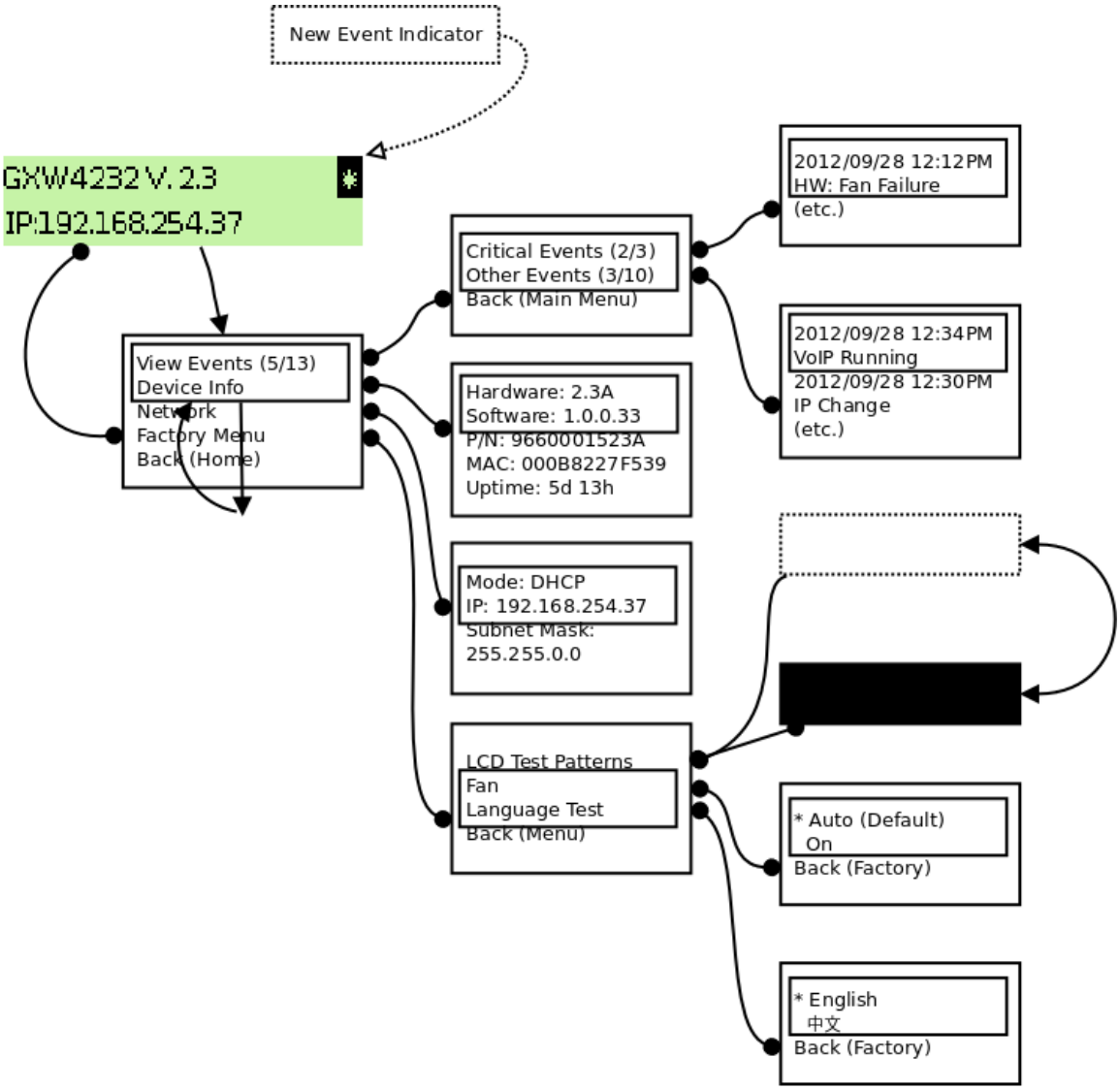


Figure 3: LCD Menu

BASIC OPERATIONS

Understanding GXW Voice Prompts

GXW42XX has a built-in voice prompt menu for simple device configuration. To enter the voice prompt menu, press *** on the standard analog phone which is connected to device's FXS port.

Table 5: Definitions of the GXW Voice Prompts

Menu	Voice Prompt	User's Options
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter "001 – 005,007, 010, 013 - 017,047, 086 or 099" Menu option Enter "9" for confirming an option
001	"DHCP Mode", "PPPoE Mode" or "Static IP Mode"	Enter '9' to toggle the selection If user selects "Static IP Mode", user need configure all the IP address information through menu 002 to 005. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.
002	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
003	"Subnet " + IP address	Same as Menu option 002
004	"Gateway " + IP address	Same as Menu option 002
005	"DNS Server " + IP address	Same as Menu option 002
007	Preferred Vocoder	Enter "9" to go to the next selection in the list: • PCMU • PCMA • iLBC • G-726 • G-723 • G-722 • G-729
010	"MAC Address"	Announces the Mac address of the unit.

013	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12-digit new IP address.
014	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12-digit new IP address.
015	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter "9" to toggle between **TFTP** and **HTTP**
016	Firmware Version	Firmware version information.
017	Firmware Upgrade	Firmware upgrade mode. Enter "9" to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
047	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
086	Voice Mail	Number of voice mails
099	"RESET"	Enter "9" to reboot the device; or Enter MAC address to restore factory default setting (See **Restore Factory Default Setting** section)
700-748	Phone calls between different ports of the same GW42xx	GXW42XX support inter-port calling from voice menu for easy test/verification in factory "700" Ring all ports and connect to first port pick up the call "701-732" Call individual port
	"Invalid Entry"	Automatically returns to Main Menu

Five Success Tips when using the Voice Prompt

1. "*" shifts down to the next menu option
2. "#" returns to the main menu
3. "9" functions as the ENTER key in many cases to confirm an option
4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add **0** before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168**00**0**0**26. No decimal is needed).
5. Key entry cannot be deleted but the phone may prompt error once it is detected

Placing a Phone Call

Phone or Extension Numbers

1. Dial the number directly and wait for 4 seconds (To change the default value, modify the following setting - “No Key Entry Timeout”)
2. Or, you may dial the number directly and press # (Use # as dial key” must be configured in web configuration).

Examples:

1. Dial a number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both GXW42XX and other VoIP Device, have public IP addresses, or
2. Both GXW42XX and other VoIP Device are on the same LAN using private IP addresses, or
3. Both GXW42XX and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

GXW42XX supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial “*******”
2. Dial “**047**” to access the direct IP call menu
3. Enter the IP address using format ex. **192*168*0*160** after the dial tone.

Using Star Code

1. Pick up the analog phone then dial “***47**”
2. Enter the target IP address using same format as above.

Note: **NO** dial tone will be played between step 1 and 2.

Destination ports can be specified by using “*****” (encoding for “:”) followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is

***47 or Voice Prompt with option 047, then 192*168*0*160** followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:

***47 or Voice Prompt with option 047, then 192*168*0*160*5062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

Note: When completing direct IP call, the "**Use Random Port**" should set to "NO".

Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button). Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

Call Waiting

Call waiting tone (Repeated short beeps as long as the caller is still calling) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

Call Transfer

Blind Transfer

Assume that call Caller A and B are in conversation. "A" wants to *Blind Transfer* "B" to "C":

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.
4. Caller A dials ***87** then dials caller C's number, and then # (or wait for 4 seconds).
5. Caller A will hear the confirm tone. Then, A can hang up.

Note: "*Enable Call Feature*" must be set to "Yes" in web configuration page.
Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.

2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Continuous busy tone. The phone has timed out. **Note:** continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials Caller C's number followed by **#** (or wait for 4 seconds).
3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.
4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

Note: When Attended Transfer fails and A hangs up, the GXW42XX will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-Way Conferencing

The GXW42XX supports Bellcore style 3-way Conference.

Instructions for 3-way conference:
Assuming that call party A and B are in conversation. A (GXW42XX) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.
6. If A hangs up, the conference will be terminated or transfer B to C if "Transfer on Conference Hangup" set to yes

Hunting Group

This feature allows the user to setup a single SIP account on the gateway and have the ability to use all FXS ports to make/receive calls. Using this feature, all ports active in same Hunting Group will have the same phone number and incoming calls will be distributed in a Linear or Circular manner among the ports

active in that Hunting Group. The number of hunting groups is limited by the number of ports each GXW model has - i.e. each port can be its own Hunting Group. The most practical and efficient way to use Hunting Groups is to assign 2 or 3 ports to separate Hunting Groups.

One additional and popular way to use the Hunting Group feature is called "*multiplexed analog lines*". In this configuration, a legacy PBX system with 8 FXO trunks can be connected to 8 GXW 42xxports configured as a Hunting Group. The GXW can be registered to a SIP server provider using only one phone number. If the SIP service provider allows multiple calls to the same number, the GXW will allow 8 concurrent calls to the same SIP number. All office members can be reached remotely using the same phone number in a round-robin fashion.

Example Configuration of a typical Hunting Group:

1. Configure the SIP account from your VoIP Service Provider on **FXS port 1** under **FXS Ports** webpage.
2. Select **Active** under the **Hunting Group** drop box for FXS port 1.
3. For the remaining ports (say 2, 3 and 4) select **1** for **Hunting Group**. Ports 2, 3 and 4 are now active members of the hunting group associated with port 1.

This configuration will route all calls directed to FXS port 1 to ports 2, 3 and/or 4 in round robin fashion respectively *if* port 1 is busy or times out. You can configure the ring timeout on the **Profile** page.

Example configuration of a multiple Hunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #4: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"

FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"

FXS Port #7: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**7**"

…

Hunting Group 1 contains ports 1, 2, 3. Hunting Group 4 contains ports 4, 5, 6.
Hunting Group 7 contains ports 7, 8.

Please be aware, the choice of 1 for ports 2 and 3, the choice of 4 for ports 5 and 6, the choice 7 for port 8 is required to indicate that the SIP account tied to port marked as **"Active"** will be used for all members of

the same Hunting group. Needless to say, those members of the same Hunting group may not be sequential ports. In following example ports 3, 5 and 7 tied to SIP Account configured in Port #1 marked as **"Active"**, and ports 4,6,8 tied to SIP Account configured in Port #2 marked as **"Active"** as well.

Example of not sequential configuration of a multiple Hunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #2: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #4: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

FXS Port #7: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

…

FXS Port #24: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

Note: A single call directed to the SIP account will NOT result in all ports ringing at the same time. They will ring in the hunting group only. This feature is applicable to incoming calls only.

There are two types of hunting groups, Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called "serial hunting". Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile X page.

Inter-Port Calling

In some cases, a user may want to make phone calls between the phones connected to multiple ports of the same gateway when it is used as a standalone unit, without the use of a SIP server. This feature will also be applicable when the gateway is used with Hunting Groups and is registered to SIP server only with one master number. In such cases, users still will be able to make inter-port calls by using the IVR feature.

For example, on the GXW42XX inter-port calling is achieved by dialing *** and 7 plus two extra digits corresponding to the port number. For example, the user connected to port 1 can be reached by dialing *** and 701; the user connected to port 24 can be reached by dialing *** 724.

Sending and Receiving Fax

GXW42XX supports fax in two modes: 1) T.38 (Fax over IP) and 2) Fax Pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

CALL FEATURES

GXW42XX supports the traditional telephony features available in a PBX as well as additional advanced telephony features.

Table 6: Call Features Table (Star Code)

Key	Call Features
***02**	**Forcing a Codec** (per call): *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *0272616 (G726-r16), *0272624 (G724-r24), *0272632 (G726-r32), *0272640 (G726-r40), *027201 (iLBC)
***03**	**Disable LEC** (pe call) Dial "*03" +" number". **No** dial tone is played in the middle.
***16**	**Enable SRTP**
***17**	**Disable SRTP**
***30**	**Block CallerID** (for all-config change)
***31**	**Send CallerID** (for all-config change)
***67**	**Block CallerID** (per call)
***82**	**Send CallerID** (per call)
***47**	**Direct IP Calling.** Dial "*47" + "IP address". **No** dial tone will be played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all-config change)
***51**	**Enable Call Waiting** (for all-config change)
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (Per Call)
***71**	**Enable Call Waiting** (Per Call)
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward:** Dial "*73" and wait for dial tone, then hang up.
***74**	**Enable Paging Call:** Dial "*74" and then the destination phone number you want to activate in Paging mode.

***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls will be rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls will be accepted.
***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward**: dial "*91". Wait for dial tone. Hang up.
***92**	**Delayed Call Forward:** Dial "*92" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward:** Dial "*93" for a dial tone, then hang up.
***98**	**Play registration ID:** Dial "*98", the registration ID will be announced.
***99**	**Provision start:** Dial "*99", the provisioning process will be triggered.
Flash/Hook	If user hears call waiting beep, flash/hook will switch to the new incoming call. Also used to switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

CONFIGURATION GUIDE

Configuring GXW42XX via Voice Prompt

DHCP MODE

Select voice menu option 001 to enable GXW42XX to use DHCP.

STATIC IP MODE

Select voice menu option 001 to enable GXW42XX to use STATIC IP mode, then use option 002, 003, 004, 005 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

PPPOE MODE

Select voice menu option 001 to enable GXW42XX to use PPPoE mode.

FIRMWARE SERVER IP ADDRESS

Select voice menu option 013 to configure the IP address of the firmware server.

CONFIGURATION SERVER IP ADDRESS

Select voice menu option 014 to configure the IP address of the configuration server.

UPGRADE PROTOCOL

Select voice menu option 015 to choose firmware and configuration upgrade protocol. User can choose between TFTP, HTTP, and HTTPS.

FIRMWARE UPGRADE MODE

Select voice menu option 017 to choose firmware upgrade mode among the following three options:

1) Always check, 2) check when pre/suffix changes, and 3) never upgrade

Configuring GXW42XX with Web Browser

The GXW42XX series gateway has an embedded Web server that allows users to configure the GXW42XX through a web browser. It has language support to English, Chinese, French, and Spanish.

Access the Web Configuration Menu

The GXW42XX HTML configuration menu can be accessed via Ethernet port:

To access the HTML configuration menu from the Ethernet port:

1. Follow table 4 to find the Ethernet port IP address.
2. Open a web browser, type in the IP address – for example: http://GXW42XX -IP-Address (the GXW42XX IP-Address is the Ethernet IP address for the GXW42XX).

Note:
The IVR announces 12 digits IP address, you need to strip out the leading "0" in the IP address. For ex. IP address: 192.168.001.014, you need to type in http://192.168.1.14 in the web browser.

Once the HTTP request is entered and sent from a web browser, the user will see a log-in screen. There are two default passwords for the login page:

User Level	Username	Password	Web pages allowed
End User Level	user	123	Only Status and Basic Settings
Administrator Level	admin	admin	Browse all pages
Viewer Level	viewer	viewer	View all pages. Not allowed to modify the content.

The password is case sensitive with maximum length of 25 characters. Only an administrator can access the "ADVANCED SETTINGS", "Profile 1~4", and "FXS Ports" configuration pages. Please reference the GUI pages using the following link:
http://www.grandstream.com/sites/default/files/Resources/gxw42xx_gui.zip

Important Settings

The end-user must configure the following settings according to the local environment.
Note: Most settings on the web configuration pages are set to the default values.

NAT Settings

If you plan to keep the gateway within a *private network* behind a firewall, we recommend using STUN Server. The following three (3) settings are useful in the STUN Server scenario:

1. **STUN Server** (under Advanced Settings webpage)

Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.

2. **NAT Traversal** (under the Profile web pages)
   Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

DTMF Settings are in Profile pages.

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

You can enable set priority of DTMF methods according to your preference, from Priority 1 to 3. This setting should be based on your server DTMF setting.

Preferred Vocoder (Codec)

The GXW42XX supports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

- PCMU/A (or G711μ/a)
- G729
- G723
- G726 (16/24/32/40)
- G722
- iLBC
- AAL2 (all G726)

Definitions

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **STATUS:** Displays the network status, account status, software version and MAC-address of the phone
- **MAINTENANCE:** Basic settings such as basic network, date and time settings, and web/SSH access settings can be set here.

- **PROFILE - AUDIO SETTINGS:** DTMF, Vocoder, and Analog Line settings can be configured here for each port.

Additional functions available to administrators are:

- **MAINTENANCE:** Full settings for network, upgrade/provisioning, TR-069, Security and Syslog.
- **ADVANCED SETTINGS:** To set advanced Ring Tongs, FXO Failover, and System Features.
- **PROFILE X:** To configure each of the SIP accounts.
- **FXS PORTS:** To configure each of the FXS ports and Hunting Groups etc.

TABLE 7: STATUS

System Info	
Product Model	Contains the product model info.
Part Number	Product Part Number.
Software Version	***Program:*** This is the main software release. *This number is always used for firmware upgrade.* Current version is 1.0.5.43. ***Boot:*** current version is 1.0.5.1 ***Core:*** current version is 1.0.5.14 ***Base:*** current version is 1.0.5.36 ***CPE:*** current version is 1.0.1.90
System Up Time	Shows system uptime since the last reboot.
System Time	The time according to NTP server.
Service Status	Shows the status of the VOIP applications.
Network Status	
MAC Address	The device ID in hexadecimal format. This is needed for Internet Service Provider troubleshooting**.** The MAC address will be used for provisioning and can be found on the label on original box and on the label located on the bottom panel of the device.
IP Address Mode	Shows the current IP mode.
IP Address	Shows IP address of GXW42XX.
Subnet Mask	Shows Subnet Mask of GXW42XX.

Gateway	Shows Default Gateway of GXW42XX.
DNS Server	Shows DNS Server of GXW42XX.
NAT Traversal	Shows type of NAT the GXW42XX is connected to via its WAN port. It is based on STUN protocol.

Port Status

Displays relevant information regarding the individual FXS ports. Example:

Port	Hook	SIP Registration	DND	Forward	Busy Forward	Delayed Forward
FXS1	On Hook	Registered	No	613		
FXS2	On Hook	Registered	No		614	
FXS3	On Hook	Not Registered	No			
FXS4	On Hook	Registered	Yes			615
FXS…	…	…				

TABLE 8: MAINTENANCE

Network Settings	
IP Address Mode	Choose how the IP address obtained on the phone.
Preferred DNS Server	Enter the preferred DNS server that should be used for DHCP and PPPoE.
DHCP Settings	
Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by Internet Service Providers.
DHCP Domain	Specifies the DHCP Domain. This value is optional but may be required by Internet Service Providers.
Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class ID.
PPPoE Settings	
PPPoE Account ID	Configures PPPoE Account ID.

PPPoE Password	Configures PPPoE Password.
PPPoE Service Name	Specifies a name for PPPoE.
Static IP Settings	
IP Address	Configures static IP Address, Subnet Mask, Gateway, DNS Server 1, and DNS Server 2.
Subnet Mask	Configures Subnet Mask
Gateway	Configures Gateway address.
DNS Server 1	Configures primary DNS Server.
DNS Server 2	Configures secondary DNS Server.
Network settings for Management Interface	
Enable management interface	An interface for HTTP/HTTPS/SSH, the management interface will be a VLAN interface if management VLAN is configured and different to service VLAN, otherwise, a sub interface will be created on service interface. Sub interface can only be configured with a static IP address without the default gateway if both management and service interface are using the same VLAN the management interface will automatically use the gateway of the service interface.
Management access	Allows to choose which interface can be reached for HTTP/HTTPS/SSH
IP Address Mode for Management Interface	Allow to choose how the IP address will be obtained on the gateway for management interface.
LLDP	
Enable LLDP	Configure to enable/disable the LLDP (Link Layer Discovery Protocol) service.
Layer 2 QoS Settings	
Layer 2 QoS 802.1Q/VLAN Tag	Value used for layer 2 QoS 802.1Q/VLAN Tag. Default setting is **0**.

Layer 2 QoS 802.1p Priority Value for SIP signaling	Value used for layer 2 802.1p Priority Value for SIP signaling. Default setting is **0**.
Layer 2 QoS 802.1p Priority Value for RTP media	Value used for layer 2 802.1p Priority Value for RTP media. Default setting is **0**.
STUN Settings	
Use STUN	Enable STUN. Default is **No**.
STUN server	The IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
Number of STUN Response Misses Allowed	The Number of STUN response misses allowed before restarting DHCP. The minimum is 3 misses.
Keep-Alive Interval	Specifies in seconds how often the phone sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. The default is 20 seconds.
Firewall Setting	
Black List for WAN Side Port	allows users to block access for specific IP address.
Reply to ICMP	This feature allows users to Enable/Disable ICMP response
Upgrade and Provisioning	
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.
Firmware Upgrade and Provisioning	Specifies how firmware upgrading and provisioning request to be sent. There are three options to choose from: “Always Check for New Firmware”, “Check New Firmware only when F/W pre/suffix changes”, and “Always Skip the Firmware Check”.
XML Config File Password	The password used for encrypting the XML configuration file using Open SSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name needed to authenticate with the HTTP/HTTPS server.
HTTP/HTTPS Password	The password needed to authenticate with the HTTP/HTTPS server.

Always send HTTP Basic Authentication Information	Default is **No**. If set to Yes, device will send configured user name and password within HTTP request before server sends authentication challenge.
Upgrade via	Allows users to choose the firmware upgrade method via TFTP, HTTP or HTTPS.
Firmware Server Path	IP address or domain name of firmware server. That URL of the server that hosts the firmware release. The default server is: **fm.grandstream.com/gs**
Config Server Path	IP address or domain name of configuration server. The server hosts a copy of the configuration file to be installed on the gateway. The default server is: **fm.grandstream.com/gs**
Firmware File Prefix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching prefix will be downloaded.
Firmware File Postfix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching postfix will be downloaded.
Config File Prefix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching prefix will be downloaded.
Config File Postfix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching postfix will be downloaded.
Allow DHCP Option 43 and Option 66 to Override Server	If set to "Yes", configuration and upgrade server's information can be obtained using DHCP option 66 from DHCP server. This option specifies the URL of the TFTP server. **Note:** If DHCP Option 66 is enabled, the gateway will attempt downloading a configuration file from the server URL provided by DHCP, even though **Config Server Path** is left blank.
3CX Auto Provision	Default is **No**. If set to **Yes**, phone will multicast SUBSCRIBE for provision if this feature is enabled.

Automatic Upgrade	Choose "*Yes*" to enable automatic upgrade and provisioning. When set to No, GXW42XX will only do upgrade once at boot up. When "Check every day" or "Check every week" is checked, user can specify "Hour of the day (0-23)" or "Day of the week (0-6)". Default time is Monday 1AM.
Randomized Automatic Upgrade	With this feature enabled, device will specify a random time to upgrade the device within the range of hours of the day or postpone the upgrade every X minute(s) by random 1 to X minute(s)
Hour of the Day (0-23)	Define the hour of the day to check HTTP/TFTP server for firmware upgrades or configuration files changes
Day of the Week (0-6)	Define the days of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes
Authenticate Conf File	If set to Yes, configuration file is authenticated before being accepted. This protects the configuration from unauthorized modifications.
Firmware Key	For firmware encryption. It should be 32-digit in Hexadecimal Representation. End user should keep it blank.
Upload Device Configuration	Upload the Config file to reload settings.
Upload Device Firmware	Upload .bin file to the device to upgrade the firmware.
Device Configuration (Text format)	Click to download the device configuration file in .txt format.
Device Configuration (XML format)	Click to download the device configuration file in .xml format.
Web/SSH Access	
HTTP Web Port	By default, HTTP uses port 80. This field is for customizable web port.
HTTPS Web Port	By default, HTTP uses port 443. This field is for customizable web port.
Web Access Mode	Specify Web Access Protocol. HTTP or HTTPS. Default is HTTP.
Disable SSH	If set to Yes, SSH access will be disabled. Default is **No**

Web Session Timeout	Configure timer to logout web session during idle. Default is 10 min. Range is 2-60 min
Web access Attempt Limit	Configure attempt limit before lockout. Default is 5. Range is 1-10
Lockout Time Interval	If login attempt failed 5 times, login would be locked out for the time length. (Default 15 mins. Range 1-15 min).
Disable User Level Web Access	If set to yes, User will not be able to access web UI
Disable Viewer Level Web Access	If set to yes, Viewer will not be able to access web UI
Viewer Password	Set new password for web GUI access as Viewer. This field is case sensitive with a maximum length of 30 characters.
User Password	Set new password for web GUI access as User. This field is case sensitive with a maximum length of 30 characters.
Admin Password	Set new password for web GUI access as Admin. This field is case sensitive with a maximum length of 30 characters.
Access Control Lists	White list for WAN side: only IP in this list can access web and SSH. Black list for WAN side: IP in this list can't access web and SSH. Multiple IPs are supported and need to be separated by “space”. Example: 192.168.5.222 192.168.5.223 192.168.7.0/24
TR-069	
Enable TR-069	Enable TR-069 service.
ACS URL	TR-069 Auto Configuration Servers URL (e.g., http://acs.mycompany.com, or IP address).
ACS Username	User specify the ACS Username.
ACS Password	User specify the ACS password.
Periodic Inform Enable	Default is **No**. If set to Yes, device will send inform packets to the ACS
Periodic Inform Interval	Frequency that the inform packets will be sent out to the ACS
Connection Request Username	The user name for the TR-069 Auto Configuration Server to connect to the phone.

Connection Request Password	The password for the TR-069 Auto Configuration Server to connect to the phone.
SNMP	
Enable SNMP	Default is **No**.
Trap Community	Name of SNMP trap community.
IP Address	IP Address of trap destination
Port	Listening Port of SNMP daemon (Default 161).
Version	Choose between (Version 1 or Version 2c).
Trap Interval	Time interval between traps (Default is 5).
Download MIB	Download the Management information base file.
RADIUS	
RADIUS for Web Access Authentication	
Enable RADIUS Web Access Control	Default is No
Action upon RADIUS Auth Server Error	Choose action upon RADIUS server error. Default is Authenticate Locally
RADIUS Auth Server Address	Address of RADIUS Auth server
RADIUS Auth Server Port	Port of RADIUS Auth server
RADIUS Shared Secret	Set RADIUS shared secret
RADIUS VSA Vendor ID	Configure RADIUS VSA Vendor IS to match RADIUS server's configuration. Default is 42397 for Grandstream Networks Inc
RADIUS VSA Access Level Attribute	Configure RADIUS VSA Access Level Attribute to match RADIUS server's configuration. Incorrect setting would cause RADIUS authenticate fail
RADIUS for call	
Primary RADIUS Server	Set Primary RADIUS Server address.

Primary RADIUS Authentication Port	Default is 1812.
Primary RADIUS Account Port	Default is 1813.
Primary RADIUS Server Secret	Set Primary RADIUS Server Secret
Secondary RADIUS Server	Set Secondary RADIUS Server address.
Secondary RADIUS Authentication Port	Default is 1812.
Secondary RADIUS Account Port	Default is 1813.
Secondary RADIUS Sever Secret	Set Secondary RADIUS Server Secret
RADIUS Timeout	Default is 2.
RADIUS Retry	Default is 3.
Security Settings	
SIP TLS Settings	
SIP TLS Certificate	The GXW42XX series supports SIP over TLS. It has built-in private key and SSL certificate. The user specified SSL certificate used for SIP over TLS is in X.509 format.
SIP TLS Private Key	You may also customize the SIP TLS Private Key. The user specified SIP TLS private key used for SIP over TLS is in X.509 format.
SIP TLS Private Key Password	SSL Private key password used for SIP Transport in TLS/TCP.
Custom Certificate	This feature allows users to upload to this device their own certificate signed by custom CA certificate to manage client authentication.

Validate Server Certificates	This feature allows users to validate server certificate with our trusted list of TLS connections. Default is enabled. The device needs to reboot after changing the setting.
Trusted CA	
Trusted CA Certificate (A, B, C, D)	The certificate entered here will be accepted as valid CA for authenticating the server TLS certificate.
Load CA Certificates	This feature specifies which CA Certificates the gateway needs to trust. Following options are available: • **Built-in trusted certificates**: The gateway will verify the server certificate based on the built-in trusted certificates list. • **Custom trusted certificates**: The gateway will verify the server certificate based on the custom trusted certificates list previously uploaded by the user. • **All trusted certificates**: The gateway will verify the server certificate based on the built-in and custom trusted certificates. Default is “Built-in Trusted Certificates”.
Date and Time	
NTP Server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time. An extensive list of public NTP servers can be found at http://www.ntp.org
NTP Update Interval	Defines the update interval (in minutes) to obtain the date and time from the server.
Allow DHCP Option 42 to NTP Server	Default is **Yes**. DHCP Option 42 is enabled, the NTP server can be changed. If set to **No** will disable the NTP server be changed.
Time Zone	Configures the date/time used on the phone according to the specified time zone.
Self-Defined Time Zone	This parameter allows the users to define their own time zone. For syntax and examples, please refer to user manual.
Allow DHCP Option 2 to override time zone	Default is **Yes**. DHCP Option 2 is enabled, the time zone can be offset. If set to **No**, it will disable the time zone be offset.

Syslog	
Syslog Server	The IP address or URL of System log server. The server collects system log information from the device.
Syslog Level	Select the GXW42XX to report the log level. Default is **NONE**. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Print SIP in Syslog	Enable or disable printing of full SIP messages in Syslog.
Save Syslog	Save Syslog to device. 3 rotate files save syslog. Each file could save at maximum 3 MB

TABLE 9: ADVANCED SETTINGS

Ring Tones	
System Ring Cadence	Configuration option for all FXS ports ring cadence for all incoming calls. (Syntax: c=on1/off1-on2/off2-on3/off3; [...]) Default is set to c=2000/4000; (US standards)

Call Progress Tones	Using these settings, user can configure tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing (ON time in ms) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. • "Dial tone" • "Ring back tone" • "Busy tone" • "Reorder tone" • "Confirmation tone" • "Call-Waiting tone" • "Prompt Dial tone" • "Conference Party hangup tone" Please refer the document below to determine your local call progress tones: http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
FXO Failover	
Failover to FXO Gateway	Enable or disable the Failover FXO Gateway.
FXO Gateway IP	IP Address or URI of the FXO gateway.
System Features	
Disable Direct IP Call	Disables the Direct IP Call function. Default is "**No**". If set to "**Yes**" direct IP-to-IP calling will not be supported.
Inter-port Calling	With this feature enabled, when the FXS port is unregistered, users can use the same extensions of the FXS port for inter-port dialing instead of dialing ***7xx.
Disable SIP NOTIFY Authentication	Disables challenging SIP NOTIFY reboot and resync messages.
Disable Voice Prompt	Disables the voice prompt configuration. Default is "**No**". If set to "**Yes**" accessing integrated voice menu will be impossible.
IVR Language	Choose English, Chinese, or Spanish.

Upload/Delete Language Pack	Upload/delete language pack.
Display Language	Choose language for LCD display.
Prompt Dial Tone Code	Simulates an analog PBX where a code is required to dial an outside line.
Country Specific Deployment	Configure the country specific deployment.
Automatic Reboot	Default is **No**. When "Yes, reboot every day" or "Yes, reboot every week" is checked, user can specify "Hour of the day (0-23)" or "Day of the week (0-6)". Default time is Monday 1AM.
LED Pattern	This feature allows users to customize LED patterns. There are three patterns to choose from: • **Pattern A:** Onhook: OFF; Offhook: On; Voicemail: Blink • **Pattern B:** Onhook: On; Offhook: Off; Voicemail: Blink if phone onhook • **Pattern C:** Onhook and Registered: On; Offhook: Blink; Voicemail: Blink 1sec ON/1sec OFF

TABLE 10: PROFILES

General Settings	
Profile Active	When set to Yes, the SIP Profile is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Failover SIP Server	Failover SIP Server's IP address or Domain name provided by VoIP Service provider. This server will be used if the Primary SIP server becomes unavailable.
Prefer Primary SIP Server	If set to yes it will register to Primary Server if registration with Failover server expires Default is No.

Primary Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW42XX for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
Backup Outbound Proxy	Backup Outbound Proxy which will be used when the primary outbound proxy cannot be connected
Prefer Primary Outbound Proxy	If yes, the profile will re-register via the primary outbound proxy when registration expires.
Network Settings	
Layer 3 QoS Settings	Configure the Diff-Serv value for SIP and RTP. Default is: SIP Diff-Serv **24** RTP Diff-Serv **46**
DNS Mode	One from the 3 modes available for "DNS Mode" configuration: • **A Record** (for resolving IP Address of target according to domain name) • **SRV** (DNS SRV resource records indicates how to find services for various protocols) • **NAPTR/SRV** (Naming Authority Pointer according to RFC 2915) • **Use Configured IP** (If selected, please fill in Primary IP, Backup IP 1 and Backup IP 2.) One mode can be chosen for the client to look up server. The default value is "A Record"
DNS SRV Use Registered IP	If yes, under DNS SRV mode, use Registered IP as INVITE destination instead of using high priority IP from resolved IP list.
Primary IP	Configures the primary IP address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
Backup IP 1	Configures the backup IP 1 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.

Backup IP 2	Configures the backup IP 2 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
NAT Traversal	This parameter defines whether the GXW42XX NAT traversal mechanism is activated or not. If activated (by choosing “STUN”) and a STUN server is also specified, then the GXW42XX performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXW42XX will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to “STUN” with no specified STUN server, the GXW42XX will periodically (every 20 seconds) send a blank UDP packet (with no payload data) to the SIP server to keep the “hole” on the NAT open.
Use NAT IP	The NAT IP address used in SIP/SDP messages. It should ONLY be used if required by your ITSP.
Proxy-Require	A SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall.
SIP Settings - Basic Settings	
SIP transport	User can select UDP or TCP or TLS. Please make sure you’re SIP Server or network environment supports SIP over the selected transport method. Default is UDP.
SIP Registration	This parameter controls whether the GXW42XX needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is **No**. If set to “Yes”, the SIP user’s registration information is cleared on reboot.
Add Auth Header on Initial REGISTER	Default is **No**. If set to “Yes”, an Authentication Header with blank nonce will be added in the initial REGISTER.
Outgoing Calls Without Registration	If set to “Yes,” user can place outgoing calls even when not registered (if allowed by Internet Telephone Service Provider) but is unable to receive incoming calls. Any port, member of a Hunting Group that is not registered with a SIP

	account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port. Default is **No**. For example: Port 1, 3 and 5 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 3 and 5 are not registered. Ports 3 and 5 will be able to place outbound calls using the SIP account of port 1, even if ***Outgoing Call without Registration*** is set to No.
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GXW42XX to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
SIP Registration Failure Retry Wait Time	Allows the user to specify the time frequency (in seconds) for the GXW42XX to re-register after registration failure. The default interval is 20 seconds. The maximum interval is 3600seconds (1 hour).
SIP Registration Failure Retry Wait Time upon 403 Forbidden	Default is **1200** sec. Specifies the interval to retry registration if the process is failed due to 403 Forbidden. Valid range is 0 to 3600 in second. 0 second means stop retry registration.
Reregister Before Expiration	Determines how many seconds before the previous registration expires that the port should reregister.
Local SIP Port	Defines the local SIP port the GXW42XX will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Use Random SIP Port	Default is **No**. If set to Yes, the device will pick randomly-generated SIP ports. This is usually necessary when multiple GXW42XX /HT50X are behind the same NAT.
Local RTP port	Defines the local RTP port used to listen and transmit RTP packets. The default value is 50000 for Profile 1, 51000 for Profile 2, 52000 for Profile 3 and 53000 for Profile 4.
SIP T1 Timeout	T1 is an estimate of the round-trip time between the client and server transactions. If the network latency is high, select larger value for more reliable usage.
SIP T2 Timeout	Maximum retransmission interval for non-INVITE requests and INVITE responses.

Remove OBP from Route Header	Default is **No**. If set to Yes, the Outbound Proxy will be removed from the route header.
Support SIP Instance ID	Default is **Yes**. If set to Yes, the contact header in REGISTER request will contain SIP Instance ID as defined in IETF SIP Outbound draft.
Hold Target Before Refer	Default is **Yes**. Select whether the gateway will send ReINVITE to hold transfer target before sending REFER to transferee.
Refer-To Use Target Contact	Default is **No**. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
SUBSCRIBE for MWI	Default is **No**. When set to Yes, a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Enable 100rel	Enables the use of PRACK (Provisional Acknowledgment) method.
TEL URI	The default setting is "Disabled". If the phone has an assigned PSTN Number, this field should be set to "User=Phone" then a "User=Phone" parameter will be attached to the "From header" in the SIP request to indicate the E.164 number. If server supports TEL URI format, then this option needs to be selected.
Use Request Routing ID in SIP Headers	Default is **No**. If set to Yes, device will use the configured [Request URI Routing ID] in the SIP Header. This option is usually used under a SIP trunk account's configuration.
Do Not Escape '#' as %23 in SIP URI	If set to "**Yes**", device will use '#' instead of %23 in the send URI.
Disable Multiple m Line in SDP	Default is **No**. If set to Yes, device will send only one m line in SDP, regardless how many m field in the incoming SDP.
Use Privacy Header	If set to **Default**, it will only add Privacy or PPI header when special feature is not **Telkom SA** or **CBCOM.**
Use P-Preferred-Identity Header	If set to **Default**, it will only add Privacy or PPI header when special feature is not **Telkom SA** or **CBCOM.**
SIP REGISTER Contact Header Uses	Default is **LAN Address**. If set to **WAN Address**, device will detect its WAN address and use it in SIP REGISTER Contact Header.

Caller ID Display	When set to “Auto”, the phone will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP invite. When set to “Disabled”, All incoming calls are displayed with “Unavailable”.
SIP Settings - Session Timer	
Session Expiration	The session timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. When the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is **180 seconds**.
Min-SE	The minimum session expiration (in seconds). The default value is **90 seconds**.
Caller Request Timer	If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes” the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting “Yes” the phone will use session timer even if the remote party does not support this feature. Selecting “No” will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
SIP Settings - Security Settings	
Validate Incoming Messages	Default is **No**. Defines whether the incoming messages will be validated.

Check SIP User ID for Incoming INVITE	Default is **No**. If set to **Yes**, SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn't match the phone's SIP User ID, the call will be rejected. Direct IP calling will also be disabled.
Accept Incoming SIP from Proxy Only	Default is **No**. If set to **Yes**, SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn't match the phone's SIP User ID, the call will be rejected. Direct IP calling will also be disabled.
Authenticate Incoming INVITE	Default is **No**. If set to **Yes**, the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response.
Authenticate server certificate domain	Default is **No**. If this is set to **Yes**, device will check the server TLS certificate to ensure that the Common Name matches the configured SIP server
Authenticate server certificate chain	Default is **No**. If this is set to **Yes**, device will check the server TLS certificate to ensure that it is authorized by a known Certificate Authority
Fax Settings	
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Fax Tone Detection Mode	Default is **Callee**. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Send Re-INVITE After Fax Completion	Default is **No**, If set to “Yes”, device will send an INVITE with audio vocoders upon competition of Fax to continue session in audio only.
Send Re-INVITE After Fax Tone	If set to “Yes”, device will send a Re-INVITE after Fax tone is detected, disabling will only work under Broadsoft feature.
Enable Silence Detection for Fax Disconnect	For fax machines that do not send a Disconnect when fax is done. This option Enables/Disables the detection of silence in order to know the fax has finished. The silence period is non-configurable and fixed to 7 seconds.
Audio Settings	
Preferred DTMF method (in listed order)	The GXW42XX supports up to 3 different DTMF methods including in-audio, via RTP (RFC2833) and via Sip Info. The user can configure DTMF method in a priority list.

DTMF-RELAY Tag Respect SIP INFO	Default is **No**. If set to No, the DTMF-relay tag in Accept Header is always added in outbound INVITE. If set to Yes, it depends on if SIP INFO is chosen in Preferred DTMF Method.
Disable DTMF Negotiation	Default is **No**. If set to yes, use above DTMF order without negotiation
Enable Fast RFC2833	Default is **No**. DTMF will use 240 ms timing. If set to yes, DTMF will 100 ms instead.
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
Preferred Vocoder	The GXW42XX supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B, iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8".
Voice Frames per TX	Number of the frame size when it transmits. Default is **2**, from 1 -4 for G711/G726/G729
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
G726-32 Packing Mode	Choose the packing mode for G726-32
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Default value is **97**. Defines payload type for iLBC. The valid range is between 96 and 127.
AAL2-G726-16 Payload type	Default value is**100**. Range is from 96 to 127.
AAL2-G726-24 Payload type	Default value is **99**. Range is from 96 to 127.
AAL2-G726-32 Payload type	Default value is **104**. Range is from 96 to 127.
AAL2-G726-40 Payload type	Default value is **103**. Range is from 96 to 127.
Use First Matching Vocoder in 200OK SDP	Default is **No**. If set to "Yes", device will include only the first match vocoder in its 200OK response, otherwise it will include all match vocoders in same order received in INVITE.

SRTP Mode	Default is **Disabled**. Other options are **Enabled but not forced**, and **Enabled and forced**. It uses SDP Security Description to exchange key. Please refer SDES: http://www.apps.ietf.org/rfc/rfc4568.html SRTP: http://www.apps.ietf.org/rfc/rfc3711.html
Crypto Life Time	Default setting is **Yes**. Enable or disable the crypto life time when using SRTP. If users set to disable this option, phone does not add the crypto life time to SRTP header.
Silence Suppression (VAD)	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions. • **High** (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement • **Medium** (initial 100ms, min 20ms, max 200ms) • **Low** (initial 50ms, min 10ms, max 100ms)
RTCP	With this feature enabled, the device will send RTCP Sender Report during the conversation until it's ended.
SLIC Setting	Depends on standard phone type (and location)
Ringing Frequency	This feature allows to customize ringing frequency.
Caller ID Scheme	Select the value according to the local Telco standard where the GXW42XX is installed. Please refer to the pull-down list to select.
Polarity Reversal	Default is **No**. If set to "Yes", polarity will be reversed upon call establishment and termination.
Loop Current Disconnect	Set to Yes if the traditional PBX you are using with GXW42XX uses this method for signaling call termination. Default is **No**.
Play busy/reorder tone before Loop Current Disconnect	Default is **No**. If set to yes, it will play busy/reorder tone before loop current disconnect upon call fail.

Loop Current Disconnect duration	Default is **200**.In 100 - 10000 milliseconds range.
Enable Hook Flash	If no, flash will be treated as an on-hook event.
Hook Flash timing (Minimum)/ (Maximum)	Sets the minimum/maximum time the phone is on hook before being detected as a hook flash. The range is 40 to 2000 milliseconds
On Hook Timing	Sets the time required to detect that the phone is on hook. The range is 40 to 2000 milliseconds. Default is 400 milliseconds.
Gain	Handset volume adjustment. RX is for receiving volume (direction FXS→to analog phone), TX is for transmission volume (Analog phone→ to FXS). Default values are **0dB** for both parameters. Loudest volume: +6dB Lowest volume: -6dB.
Disable Line Echo Canceller (LEC)	Default is **No**. If set to "Yes", device will not use LEC to remove echo from a voice communication.
SLIC Setting	Depends on standard phone type (and location)
Caller ID Scheme	Select the value according to the local Telco standard where the GXW42XX is installed. Please refer to the pull-down list to select.
Call Settings	
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Dial Plan	**Dial Plan Rules**: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d

2. Grammar: x - any digit from 0-9;

    a. xx+ - at least 2 digits number;
    b. xx – at least 2 digits number;
    c. ^ - exclude;
    d. [3-5] - any digit of 3, 4, or 5;
    e. [147] - any digit 1, 4, or 7;
    f. <2=011> - replace digit 2 with 011 when dialing
    g. < =1> - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed
    h. | - or

- **Example 1:** {[369]11 | 1617xxxxxxx} –

  Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617

- **Example 2:** {^1900x+ | <=1617>xxxxxxx} –

  Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers

- **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} –

  Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing.

3. Default: Outgoing - {x+}

<u>Example of a simple dial plan used in a Home/Office in the US</u>:

{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }

Explanation of example rule (reading from left to right):

- **^1900x. -** prevents dialing any number started with 1900
- **<=1617>[2-9]xxxxxx -** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically
- **1[2-9]xx[2-9]xxxxxx |-** allows dialing to any US/Canada Number with 11 digits length
- **011[2-9]x. -** allows international calls starting with 011

[3469]11 - allow dialing special and emergency numbers 311, 411, 611

	and 911 **Note:** In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan will be: **{ [x*]+}**.
Use # as Dial Key	Allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", "#" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "#" key can be included as part of number.
Disable # as Redial key	When "Use # as dial key" is enabled, and this option is set to NO, the "#" key will function as redial key but If it's set to "Yes" "#" key will not function as redial key.
No Key Entry Timeout	Default is **4 seconds**. Call will be completed within this time interval if no additional key entry occurs.
Off-Hook Auto-Dial Delay	Determines how many seconds after off-hook to wait before autodialing the extension set under Advanced Port Settings. The range is 0 to 60 seconds.
Allow Auto-Dial Config Per Port	Allow users to configure Auto-Dial feature per port when hunting group is enabled. This feature is disabled by default.
Disable Call Waiting	Default is **No**. If set to **Yes** Call Waiting indication information will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Caller ID	Default is **No**. If set to **Yes** Call Waiting caller ID will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Disable Connected Line ID	Default is **No**. If set to **Yes**, Connected Line ID will not be displayed even received.
Disable Receiver Offhook Tone	Default is **No**. If set to **Yes**, ROH tone will not be played after off hook for 60 seconds.

Disable Reminder Ring for On-Hold Call	Default is **No**. This is to disable the Reminder Ring that is played when a call is waiting on hold and the analog phone goes on-hook.
Disable Voicemail Reminder Tone	If this feature is set to yes, the reminder tone will not be played when there is voicemail.
Disable Visual MWI	Visual message indicator is a special on-hook caller ID type message that enables and disables the message waiting light on certain phones. GXW42XX has this feature enabled by default. However, certain phones (rare) that do not support it may mistakenly treat this CID signal as an incoming call. A configuration option is needed to turn on MWI in this case.
Visual MWI Type	This is the type of signal sent to the analog phone to make it turn the lamp ON upon receiving a Voice mail. Check the phone's manual to find out what signal is supported, FSK (default) or Neon.
Transfer on Conference Hangup	Defines whether the call is transferred to the other party if the conference initiator hangs up.
Send Hook Flash Event	Default is **No**. If set to yes, flash will be sent as a DTMF event.
Flash Digit Control	Default is **No**. If set to **Yes** will overrides the default settings for call control when both channels are in use. • Flash+1 to hang up current call and resume call on hold • Flash+2 to hold current call and resume call on hold • Flash+3 for 3WC • Flash+4 for attended transfer
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
Delayed Call Forward Wait Timeout	Default value is **20 seconds**. In case this feature activated using * codes (*92 code), the call will be forwarded after this preconfigured amount of time.
Send Anonymous	If this parameter is set to "Yes", the "From" header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.

Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Hunting Group Type	Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called “serial hunting”. Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile X page.
Hunting Group Ring Timeout	Default is **20 seconds**. If call is not answered within this designated time period, the call will be forwarded to the next member of a Hunt Group.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors. Example of vendors - CBCOM, RNK.
Call Features Settings	
Enable Call Features	When enabled, Do not Disturb, Call Forward and other call features can be used via the local feature code on the phone. Otherwise, the ITSP feature code will be used. Enable All will override all individual features enable setting. **Note**: When call features on GXW42xx are enabled (UCF, DND...), if picking up the handset, GXW42xx play stutter tone to indicate that a call feature is active then the dial tone will be played after that.
SRTP Feature	Default is **Yes**, and star code is Enable-**16**, Disable-**17**.
SRTP per call Feature	Default is **Yes**, and star code is Enable-**18**, Disable-**19**.
CID Feature	Default is **Yes**, and star code is Enable-**31**, Disable-**30**.
CID per call Feature	Default is **Yes**, and star code is Enable-**82**, Disable-**67**.
Direct IP calling Feature	Default is **Yes**, and active star code is **47**.
CW Feature	Default is **Yes**, and star code is Enable-**51**, Disable-**50**.

CW per call Feature	Default is **Yes**, and star code is Enable-**71**, Disable-**70**.
Call return Feature	Default is **Yes**, and active star code is **69**.
Unconditional forward Feature	Default is **Yes**, and star code is Enable-**72**, Disable-**73**.
Busy forward Feature	Default is **Yes**, and star code is Enable-**90**, Disable-**91**.
Delayed forward Feature	Default is **Yes**, and star code is Enable-**92**, Disable-**93**.
Paging Feature	Default is **Yes**, and active star code is **74**.
DND Feature	Default is **Yes**, and star code is Enable-**78**, Disable-**79**.
Blind Transfer Feature	Default is **Yes**, and active star code is **87**.
Disable LEC per call Feature	Default is **Yes**, and active star code is **03**.
3WC Feature	**Disable Bellcore-style 3-Way Conference** default is **No**. If set to **Yes**, flash on 2nd line will not bring 2nd line in to the conference with 1st line. Instead, flash will switch between lines. **Star Code 3WC Feature** default is **Yes**. **3WC** active star code is **23**.
Provision start Feature	Default is **Yes**, and active star code is **99**.
Play registration id Feature	Default is **Yes**, and active star code is **98**.
Forced Codec Feature	Default is **Yes**, and active star code is **02**. For enabling each codec feature, default is **Yes**. Forced codec code (**02** in default case) is followed by Codec code. Default code of codec: PCMU – 7110 PCMA – 7111 G723 – 723 G729 – 729 AAL2-G728-R16 – 72616 AAL2-G728-R24 – 72624 AAL2-G728-R32 – 72632 AAL2-G728-R40 – 72640 iLBC – 7201 G722 – 722

Ring Tones	
Distinctive Ringtone	Customizes the Ring Tone 1 to 3 with associate caller ID: when selected, if caller ID is configured, then the device will ONLY use this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls using the FXS port. Distinctive ring tones can be configured not only for matching a whole number, but also for matching prefixes. In this case the symbol "x+" will be used. **For example:** if configured as 617x+, Ring Tone 1 will be used in case of call arrived from the area code 617. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page. **Note:** If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. Bellcore rings and tones are independent from custom ring tones. The custom ring tones can also be specified by alert-info header, for example *Alert-Info: ;info=ring5*
Ring Tones	Configure ring cadences according to preference.

TABLE 11: FXS PORTS

Port Settings	
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for GXW42XX to register to (SIP) servers of ITSP.

Name	Any name to identify this specific user.
Profile	Select the corresponding Profile ID (1/2/3/4)
Enable FXS (TR-069)	Enable or disable the FXS port.
Advanced Port Settings	
Offhook Auto-dial	Configure an auto dial number when Offhook.
Hunting Group	Assign this port to a certain hunting group
Request URI Routing ID	If configured, device will route the incoming call to designated port by request URI user ID in SIP INVITE.
FXO Mapping	
Map to FXO Port #	This is used only when peering with a Grandstream GXW410x. Default 1, Supported values are 1-8, meaning line 1 to 8 of the GXW410x device where the port will be mapped to.
Map to FXO Gateway IP	This is used when peering with an FXO gateway of any brand. You have to specifically mention the IP and sip port where call will be sent to.
Port	SIP port that will be annexed to the IP address above.

Saving the Configuration Changes

After user makes a change to the configuration, press the “Update” button in the Configuration Menu. The web browser will then display a message window to confirm saved changes.
Grandstream recommends reboot or power cycle the gateway after saving changes.

Rebooting from Remote

Press the “Reboot” button at the bottom of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

Configuration through a Central Server

Grandstream GXW42XX can be automatically configured from a central provisioning system.

When GXW42XX boots up, it will send TFTP or HTTP/HTTPS requests to download configuration files,

“cfg000b82xxxxxx” and “cfg00082xxxxxx.xml”, where “000b82xxxxxx” is the LAN MAC address of the GXW42XX . If the download of “cfgxxxxxxxxxxxx.xml” is not successful, the provision program will issue request a generic configuration file “cfg.xml”. Configuration file name should be in lower case letters.

The configuration data can be downloaded via TFTP or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of GXW42XX can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a central provisioning system GAPS (Grandstream Automated Provisioning System) to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer’s TFTP or HTTP/HTTPS server for further provisioning.

Grandstream also provides configuration tools (Windows and Linux/Unix version) to facilitate the task of generating device configuration files. The Grandstream configuration tools are free to end users. The configuration tools and configuration templates are available for download from http://www.grandstream.com/support/tools

SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP/HTTPS. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

Firmware Upgrade through TFTP/HTTP/HTTPS

To upgrade via TFTP or HTTP/HTTPS, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP HTTP or HTTPS, respectively. "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

e.g.firmware.mycompany.com:6688/Grandstream/1.0.4.4

e.g.firmware.grandstream.com

Notes:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.

- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at http://www.grandstream.com/support/firmware. Currently the HTTP firmware server URL is firmware.grandstream.com. For large companies, we recommend to maintain their own TFTP/ HTTP/HTTPS server for upgrade and provisioning procedures.

- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the GXW will attempt to retrieve the new image files by downloading them into the GXW420x's SRAM. During this stage, the GXW's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc.), the GXW will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.

- Firmware upgrade may take as long as 15 to 30 minutes over Internet, or just 5 minutes if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade.

- Grandstream's latest firmware is available http://www.grandstream.com/support/firmware.

  Oversea users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.

- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://www.solarwinds.com/products/freetools/free_tftp_server.aspx. Our latest official release can be downloaded from http://www.grandstream.com/y-firmware.htm.

Instructions for Local Firmware Upgrade

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW42XX device in the same LAN segment.
3. TFTP server's security settings should be changed from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Configure the Firmware Server Path with the IP address of the PC.
5. Update the change and reboot the unit.

Configuration File Download

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP/HTTPS. "Config Server Path" is the TFTP or HTTP/HTTPS server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 1 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When a Grandstream device boots up or reboots, it will issue a request for a configuration file "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". In addition, device will also requests a XML configuration file "cfgxxxxxxxxxxxx.xml". If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue a request for a generic configuration file "cfg.xml". Configuration file name should be in lower case letters.

Note:

1. The Only acceptable Config file formats to be used when provisioning are XML or binary.
2. Make sure the MAC header on the Config file is the provisioned device's MAC address or you can remove the header completely.

Firmware and Configuration File Prefix and Postfix

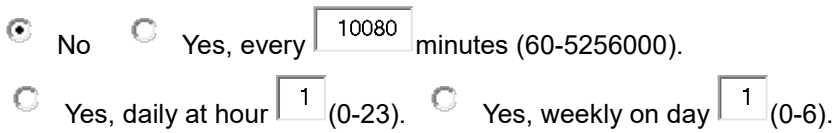
Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it possible to store ALL of the firmwares with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is selected, the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", Service Provider can use P193 to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. This allows the device periodically check whether there is any new changes need to be taken, similar to the Anti-Virus Software to upgrade the Virus Definition files. Screenshot is below:

Automatic Upgrade:



RESTORE FACTORY DEFAULT SETTING

Factory Reset

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

There are two (2) methods for resetting your unit:

Reset Button

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Insert a pin in this hole, and press for more than 4 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.
2. Enter "099" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

1. Locate the MAC address of the device. It is the 12-digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:

TABLE 12: MAC ADDRESS KEY MAPPING

Key	Mapping
0-9	0-9
A	22 (press the “2” key twice, “A” will show on the LCD)
B	222
C	2222
D	33 (press the “3” key twice, “D” will show on the LCD)
E	333
F	3333

For example: if the MAC address is 000***b***8200***e***395, it should be keyed in as “000***222***8200***333***395”.

Notes:

1. Factory Reset will be disabled if the “**Lock keypad update**” is set to “Yes”.
2. If still have difficulties to get access to the device, pushing the reset button during booting up will trigger the **RECOVERY MODE**, and device will use static IP 192.168.1.234/24. Put your PC under the same subnet, and then you will be able to open the RECOVERY MODE web page on 192.168.1.234 by your web browser.
3. While attempting to trigger the RECOVERY MODE, do not loose reset button until device enters recovery mode.
4. Admin password is needed to login RECOVERY MODE and to get the respond key if "Lock Keypad update" is set to "Yes".

EXPERIENCING THE GXW42XX

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 GXW42XX, it will be sure to bring convenience and color to both your business and personal life.

Note: 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.

This device complies with part 15 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.