![Grandstream logo - Innovative IP Voice & Video]

Grandstream Networks, Inc.

Analog IP Gateway GXW–400x
4 or 8 FXS Ports

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WELCOME

Thank you for purchasing the Grandstream GXW–400x Analog FXS IP Gateway. The GXW–400x offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GXW–400x supports popular voice codecs and is designed for full SIP compatibility and interoperability with 3rd 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 GXW–400x Overview

The new GXW–400x series has a compact and quiet design (no fans) and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The GXW–400x features 4 or 8-port FXS interface for analog telephones, dual 10M/100Mbps network ports with integrated router, PSTN life line in case of power failure, and an RS232 serial port for administration. In addition, it supports 2 SIP account profiles, caller ID for various countries/regions, T.38 fax, 1 or 2-stage dialing, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codecs including G.711 (a/u-law), G.723.1, G.726, G.728, G.729A/B/E and iLBC.

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 is contains links to Grandstream GUI Interfaces. Please remember to download these examples http://www.grandstream.com/user_manuals/GUI/GUI_GXW400x.rar for your reference.

- This document is subject to change without notice. The latest electronic version of this user manual is available for download from the following location:
  http://www.grandstream.com/user_manuals/GXW400x_userManual.pdf

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CONFIGURE YOUR GXW–400X

Connecting your GXW–400x is easy. Before you begin, please verify the contents of the GXW–400x package.

EQUIPMENT PACKAGING

Unpack and check all accessories. Equipment includes:

- one 12V universal power adapter
- one Ethernet cable
- one device unit

CONNECT THE GXW–400X

Managing the GXW and connecting the unit to the VoIP network is very simple. Follow these four (4) steps to connect your GXW to the Internet and access the unit's configuration pages.   The GUI pages will guide you through the remaining steps to set-up your gateway. Examples of the GUI Interfaces can be downloaded from: http://www.grandstream.com/user_manuals/GUI/GUI_GXW400x.rar.

FIGURE 1: DIAGRAM OF GXW–400X BACK PANEL

GXW- 400x
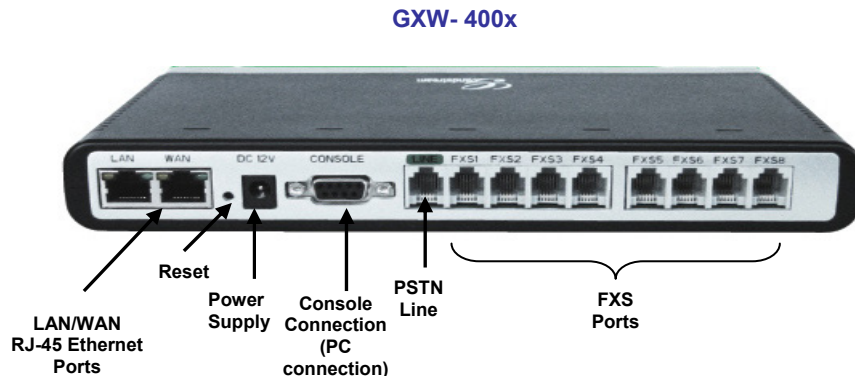
TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

LAN (or PC)	PC can be connected here as DHCP client to obtain IP Address from the internal DHCP server of the unit.
WAN (or LAN)	Connect to the internal LAN network or router.
RESET	Factory Reset button. Press for 7 seconds to reset factory default settings.
DC 12V	Power adapter connection
CONSOLE	Console connection for a PC.
FXS1 - FXS8	FXS ports to be connected to physical PSTN lines.

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

GXW- 400x

Display LEDs   FXS port status

TABLE 2: DEFINITIONS OF THE GXW DISPLAY PANEL

Power LED	Indicates Power. Remains ON when Power is connected and turned ON.
Ready LED	Remains ON after boot-up.
LAN LED	Indicates LAN (or WAN) port activity
WAN LED	Indicates PC (or LAN) port activity

LEDs 1 - 8	Indicate status of the respective FXS Ports on the back panel
	     Busy - ON (Solid Green)
	     Available - OFF

NOTE: All LEDs display green when ON. Slow blinking of FXS Port LEDs indicates Voice Mail for that port.

SAFETY COMPLIANCES

The GXW–400x is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. *Warning:* use only the power adapter included in the GXW–400x package. Using an alternative power adapter may permanently damage the unit.

WARRANTY

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

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number. Grandstream reserves the right to remedy warranty policy without prior notification.

APPLICATION DESCRIPTION

There are three scenarios where the GXW–400x series can be effectively used to enable any business to leverage the benefits of VoIP and the Internet.

EXAMPLES OF GXW-400x CONFIGURATIONS

Application One: GXW- 400x FXS Gateway Configuration / PBX Scenario, VoIP
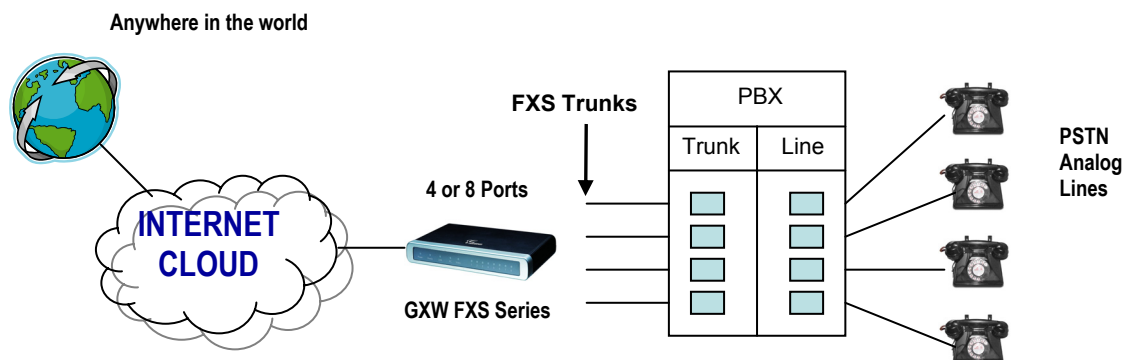
Application Two: GXW- 400x FXS Gateway Configuration / No PBX Scenario, VoIP
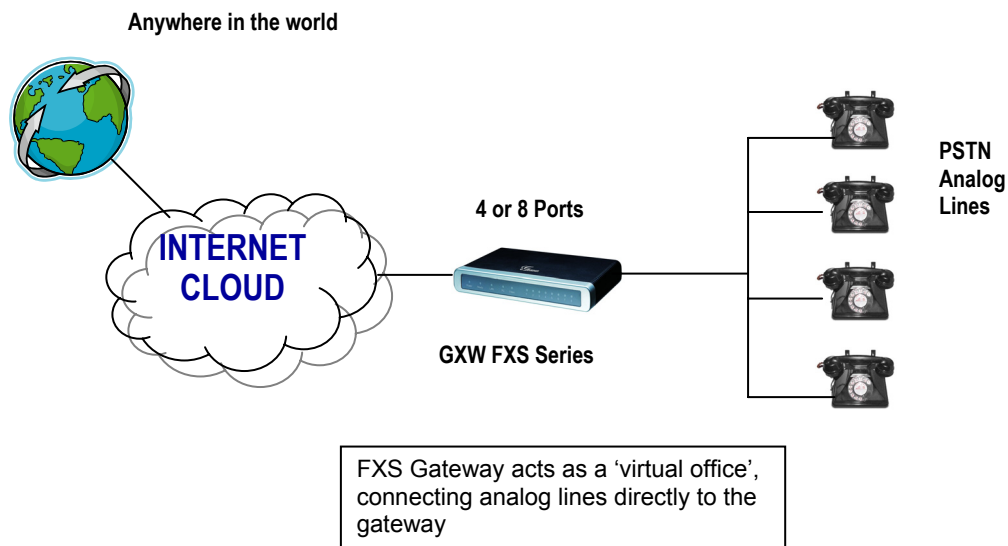
FXS Gateway acts as a 'virtual office', connecting analog lines directly to the gateway
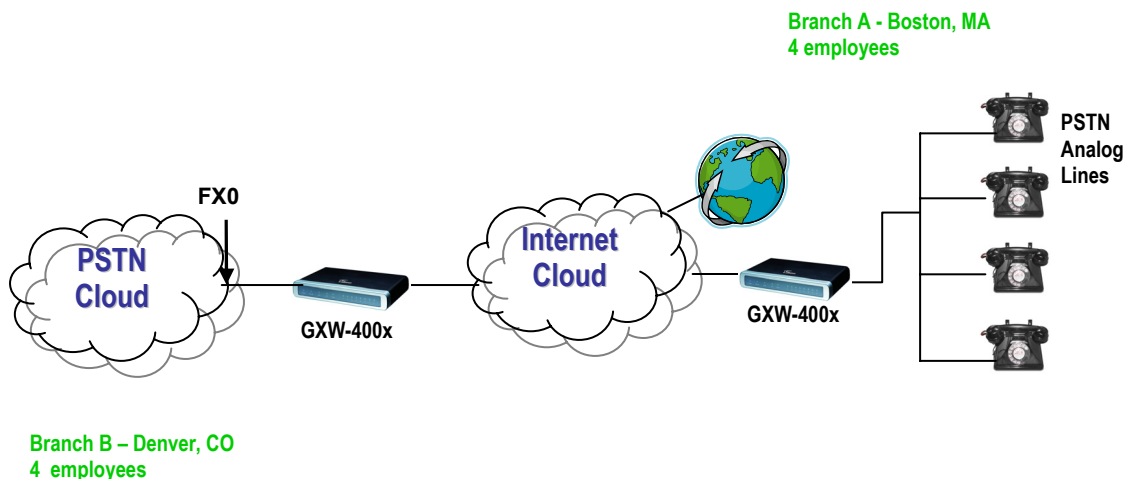
GXW–400X FEATURES

The GXW–400x is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, SoftSwitches and SIP platforms. The GXW–400x FXS series is auto-configurable, remotely manageable and scalable. There are two FXS models, the GXW-4004 and GXW-4008, each offering superb voice quality, traditional telephony functionality, easy deployment, and 4 or 8 FXS ports respectively. Each model features flexible dialing plans, PSTN failover, integrated call routing to support a pure IP network call and an external power supply.

SOFTWARE FEATURES OVERVIEW

- 4 or 8 FXS ports
- Two RJ-45 ports (switched or routed)
- Multiple SIP accounts & profiles (4 or 8 accounts / choice of 2 profiles per account)
- Supports Voice Codecs:
  G711(a/µ, Annex I & II), G723.1A, G726 (ADPCM), G728, G729 A/B/E, iLBC
- T.38 Fax
- 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

GXW– 400x FXS Analog Gateway Series

Telephone Interfaces	GXW-4004: 4 ports, 4 SIP accounts & choice of 2 profiles
	GXW-4008: 8 ports, 8 SIP accounts & choice of 2 profiles
	FXS, RJ-11
Network Interface	Two (2) 10M/100 Mbps, RJ-45
LED Indicators	Power and Line LEDs
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), AEC with NLP, Packetized Voice Protocol Unit (supports RTP/RTCP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
PSTN Fail-over	PSTN failover on power failure
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM), G.729A/B/E, 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	Yes, NAT Router or Switched Mode
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through (pending), Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP
DTMF Method	flexible DTMF transmission method, User interface of In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS (pending)
Control	TLS/SIPS
Management	Syslog support, HTTPS and telnet (pending), remote management using Web browser
Dimensions (H x W x D)	35 x 135 x 225 mm
Mounting	Rack mount, Wall mount, Desktop
Short and long haul	REN3: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes
EMC	EN55022/EN55024 and FCC part15 Class B
Safety	UL

HARDWARE SPECIFICATION

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

TABLE 4: HARDWARE SPECIFICATION OF GXW–400x

Ports	4 or 8 FXS Ports
LAN interface	2 x RJ45 10/100Mbps (switched or routed)
PSTN Port	PSTN fail-over port
LED	8 LEDs (GREEN)
PC	PC console connection
Universal Switching Power Adaptor	**Input**: 100-240V AC, 50/60Hz, 0.5A Max **Output**: 12V DC, 1.25A **UL** certified
Dimension	225mm (L) x 135mm (W) x 35mm (H)
Weight	0.29 lbs (3.5 oz)
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

USING THE WEB CONFIGURATION PAGES

QUICK END USER CONFIGURATION

1. Connect the Power to the GXW-400x unit.
2. Connect an Ethernet cable between the LAN port on GXW–400x to your PC.
3. Your PC will be assigned an IP address like 192.168.2.100
4. Launch web browser and type in http://192.168.2.1 at address of web browser. This will connect you to the web server of GXW–400x. Once this HTTP request is entered and sent from a Web browser, the GXW–400x will respond with a login screen.
5. Log-in to the GUI using the password "admin" for the Advanced Configuration Page or "123" for the Basic Configuration Page.

CONFIGURING THE GXW–400X

1. There are six different tabs (Status, Basic Settings, Advanced Settings, Profile 1, Profile 2 and FXS Ports) on the top of the screen (after login). To open each page, click on the tab.
2. Click on Profile 1 to enter your SIP Server/ SIP Proxy/Registrar information. Enter the IP Address (or FQDN) of the Server under: **SIP Server** and/or **Outbound Proxy**.
3. Click on **FXS ports** to enter the extensions or account information. You will need to fill in the following information for each extension. Once the extensions are configured, you are finished.

FXS Port	SIP User ID	Auth ID	Authentication Password	Name	Profile
1	601	601	***	Ext 1	Profile1
2	602	602	***	Ext 2	Profile1
3	603	603	***	Ext 3	Profile1
4	604	604	***	Ext 4	Profile1
5	605	605	***	Ext 5	Profile1
6	606	606	***	Ext 6	Profile1
7	607	607	***	Ext 7	Profile1
8	608	608	***	Ext 8	Profile1

4. Click **Update** after changing any setting and then **Re-boot** to confirm changes.
5. After reboot, check the Status Page to confirm the extensions are successfully registered. You can now use your standard phones connected to ports FXS1 to FXS8 to make calls.

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. **Use Random Ports** (under Advanced Settings webpage)
   This is a tricky setting. It really depends on your network settings, so set this parameter to Yes or No, whichever works. If using a Public IP address, set this parameter to No.

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

DTMF Methods

DTMF Settings are in <u>Profile pages.</u>

- DTMF in-audio

- DTMF via RTP (RFC2833)

- DTMF via SIP INFO

Enable one or more DTMF methods based on your PBX system.

Preferred VOCODER (Codec)

The GXW–400x supports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

- PCMU/A (or G711µ/a)
- G729 A/B/E
- G728
- G723
- G726-32
- iLBC

BASIC OPERATIONS

UNDERSTANDING GXW VOICE PROMPTS

GXW–400x has a stored voice prompt menu for quick browsing and simple configuration. To enter the voice prompt menu, **press ***** on the standard analog phone connected to any 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 01 – 06, 47, 86 or 99 Menu option
01	"DHCP 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 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.*
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: • PCM U / PCM A • G-723 • G-729 • iLBC • G-726-32 • G728 • G729E
10	"MAC Address"	Announces the Mac address of the unit.
12	WAN Port Web Access	Enter "9" to toggle between **enable** and **disable**
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter "9" to toggle between **TFTP** and **HTTP**
16	Firmware Version	Firmware version information.
17	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
47	"Direct IP Calling"	User will be prompted with a dial tone to dial a 12-digit IP address to make a direct IP call. (For details, see **Make a Direct IP Call**)
99	"RESET"	Enter "9" to reboot the device; or Enter MAC address to restore <u>factory default</u> setting (page 28)
	"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**0000**26. No decimal is needed).

5. Key entry can not be deleted but the phone may prompt error once it is detected

PLACING PHONE CALLS

Calling phone or extension numbers

There are currently two methods to make an extension number call:

1. Dial the numbers directly and wait for 4 (default) seconds.

2. Dial the numbers directly, and press # (assuming that "use # as dial key is selected in web configuration).

Examples:

*   To dial another extension on the same proxy, such as 1008, simply pick up the attached phone, dial 1008 and then press the # or wait for 4 seconds.

*   To dial a PSTN number such as 6266667890, you may need a prefix number followed by the phone number. Please check with your VoIP service provider for correct information. If your phone is assigned a PSTN-like number such as 6265556789, it is likely that you dial "1" prior to the number - 16266667890.

Direct IP Calls

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

1. Both GXW–400x and other VoIP Device, have public IP addresses, or
2. Both GXW–400x and other VoIP Device are on the same LAN using private IP addresses, or
3. Both GXW–400x and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP call

1. Pick up the analog phone then access the voice menu prompt by dial "***"
2. Dial "47" to access the direct IP call menu
3. Enter a 12-digit target IP address after the dial tone and voice prompt "Direct IP Calling"

Destination ports can be specified by using "***4**" (encoding for ":") followed by the port number.

NOTE: When completing direct IP call, the "**Use Random Port**" should set to "NO". You can not make direct IP calls between FXS1 to FXS2 since they are using same IP.

Examples of Direct IP Calls:

1. If the target IP address is 192.168.0.10, the dialing convention is
   Voice Prompt with option 47, then 192 168 000 010
   Press the "#" key if it is configured as a send key or wait for more than 5 seconds.

2. If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
   Voice Prompt with option 47, then 192168001020*45062 followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

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, then use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

Call Waiting

Call waiting tone (series of quick consecutive dial tones) during a conversation indicates an incoming call (if the call waiting feature is enabled). Press the flash button to place the current Caller on hold and switch to the other call. 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:

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

NOTE: "*Enable Call Feature*" must be set to "Yes" in web configuration page.

Caller A can hold the call 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 to get a 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 GXW will ring back Caller A to remind A that B is still on the call. Caller A can pick up the phone to restore conversation with B.

3-Way Conferencing

Bellcore Style 3-way Conference

The GXW–400x supports Bellcore Style 3-way Conference.

Directions for using 3-Way Conferencing:

Caller A wants to bring third Caller C into conference:

1. Caller A presses **FLASH** to receive a dial tone. (Hook Flash for older model phones)
2. Caller A dials Caller C's number then **#** (or wait for 4 seconds).
3. If Caller C answers call, then Caller A presses "flash" to bring Parties B & C into conference.
4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

CALL FEATURES

The GXW–400x 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
*30	**Block CallerID** (for all-config change)
*31	**Send CallerID** (for all-config change)
*67	**Block CallerID** (per call)
*82	**Send CallerID** (per call)
*50	**Disable Call Waiting** (for all-config change)
*51	**Enable Call Waiting** (for all-config change)
*70	**Disable Call Waiting** (Per Call)
*71	**Enable Call Waiting** (Per Call)
*72	**Unconditional Call Forward:** Dial "*72" for a dial tone. Dial 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.
*87	**Blind Transfer**
*90	**Busy Call Forward:** Dial "*90" for a dial tone. Dial 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" for a dial tone. Dial 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.
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.

PSTN PASS THROUGH/LIFE LINE

The RJ-11 line jack on the GXW–400x side functions as a pass through jack when the GXW–400x is out of power. The pass through/life line enables the user to use the analog phone for PSTN calls directly without using an access code.

T.38 FAX SUPPORT

GXW–400x 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.

END USER CONFIGURATION

The GXW FXS Series uses an imbedded web-browser to simply the configuration process. There are six (6) screenshots that the end user should become familiar with during the configuration process. Download these pages now http://www.grandstream.com/user_manuals/GUI/GUI_GXW400x.rar and review the term definitions found in Tables 7 – 11.

TABLE 7: BASIC SETTINGS PAGE DEFINITIONS

End User Password	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
IP Address	There are two modes to operate the GXW–400x : **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXW–400x acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The GXW–400x will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
Time Zone	Controls how the date/time is displayed according to the specified time zone.
Self Defined Time Zone	TBD
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the GXW.
WAN side HTTP Access	If set to "Yes", user can access the configuration page through the WAN port, instead of through the "PC" port. Warning: this configuration is less secure than default option. Default is **No**.
Reply to ICMP on WAN port	If set to "Yes", the GXW400x will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
Cloned WAN MAC Addr	Enables a specific MAC address. Set in Hex format.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is **192.168.2.1**.
DHCP IP Lease Time	Value is set in units of hours. Default value is **120 hrs** (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by GXW400x or defined in port forwarding.
Port Forwarding	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

In addition to the Basic Settings configuration page, the end user also has access to the Device Status page.

TABLE 8: STATUS PAGE DEFINITIONS

MAC Address	The device ID in HEX format. This is needed for ISP troubleshooting.
WAN IP Address	Shows LAN IP address of GXW–400x
Product Model	Contains the product model info.
Software Version	Program: This is the main software release. Boot and Loader are not changed often.
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Shows whether the PPPoE connection is running if connected to DSL modem.
Detected NAT Type	Shows type of NAT the GXW–400x is connected to via its WAN port. It is based on STUN protocol.
Port	Indicates whether the different FXS Ports are registered to the SIP server(s).

ADVANCED USER SETTINGS

Log-in to the Advanced User Configuration Page the same way as for the basic configuration page. Log-in using either of the following passwords: "**admin**" or "**123**".

FIGURE 3: SCREENSHOT OF ADVANCED USER CONFIGURATION

Grandstream Device Configuration

Password [ ]

Login

All Rights Reserved Grandstream Networks, Inc. 2005-2006
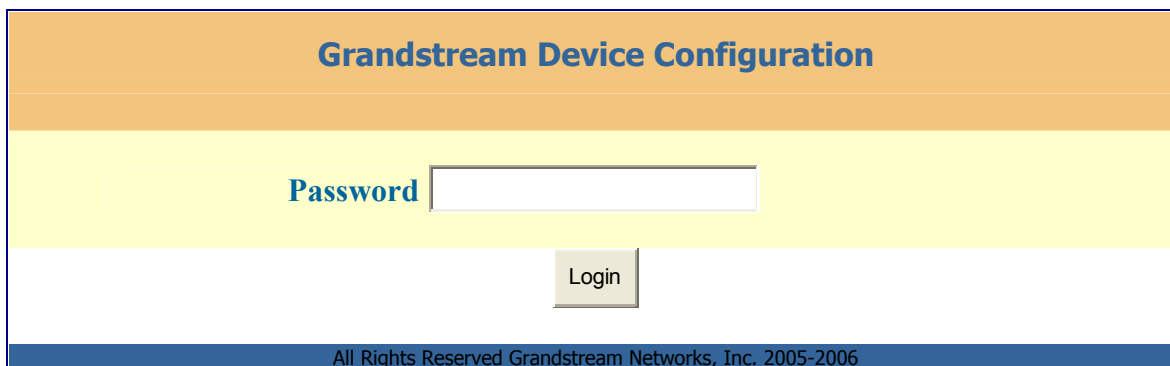
Advanced User configuration includes the end user configuration and advanced configurations including: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

Admin Password	Administrator password. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**.
Layer 2 QoS	Value used for layer 2 VLAN tag. Default setting is **blank**.
STUN Server	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the GXW–400x sends a blank UDP packet to the SIP server to keep the "hole" on the NAT open. Default is **20 seconds**.
Firmware Upgrade & Provisioning	Enables GXW–400x to download firmware or configuration file through either the TFTP or HTTP server.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the GXW–400x retrieves the new configuration file or new code image from the specified TFTP server at boot time. After 5 attempts, the system will timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image is saved into the Flash memory. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. *For example*, ttp://provisioning.mycompany.com:6688/Grandstream/1.0.0.36 "**:6688**" is the specific TCP port where the HTTP server is listening; it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, GXW–400x will only do HTTP download once at boot up.
Automatic Upgrade	Choose "*Yes*" to enable automatic upgrade and provisioning. In "Check for new firmware every" field, enter the number of days to enable GXW–400x to check the server for firmware upgrade or configuration in the defined period of days. When set to No, GXW–400x will only do upgrade once at boot up. "Always check for New Firmware." Check New Firmware only when F/W pre/suffix changes"
Authenticate Conf File	If set to Yes, config file is authenticated before acceptance. This protects the configuration from an unauthorized change.
Firmware Key	For firmware encryption. It should be 32 digit in Hexadecimal Representation. End user should keep it blank.
Lock Keypad Update	If set to "Yes", the configuration update via keypad is disabled.
NTP server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time.

Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog Level	Select the GXW–400x 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

TABLE 10: FXS PORTS CONFIGURATION DEFINITIONS

FXS Port	FXS Port Number
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 GXW–400x to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)

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.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW–400x 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.
SIP transport	User can select UDP or TCP or TLS.
NAT Traversal	This parameter defines whether the GXW–400x NAT traversal mechanism is activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW–400x 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 GXW–400x 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 "Yes" *with no specified STUN server*, the GXW–400x will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Use DNS SRV:	Default is **No**. If set to "Yes" the client will use DNS SRV to look up server.
User ID is Phone Number	If the GXW–400x has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter controls whether the GXW–400x 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.
Outgoing Call w/o Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GXW–400x 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).
Local SIP port	Defines the local SIP port the GXW–400x will listen and transmit. The default value for Account 1 is 5060. It is 5062, 5064, 5066 for Account 2, Account 3 and Account 4 respectively.
Local RTP Port	Defines the local RTP-RTCP port pair the GXW400x will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Refer to Use Target Contact	Used for call transfer, incase server requires this setting.

DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
DTMF in-audio	Send DTMF as inband (in-audio).
DTMF via RTP	Send DTMF via RTP (According to RFC 2833).
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	Default is **NO**. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is **No**. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
Off-Hook Auto Dial	Allows the user to configure a User ID or extension number to be automatically dialed upon off-hook. Only the user part of a SIP address needs to be entered here. The phone will automatically append the "@" and the host portion of the corresponding SIP address.
Proxy Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Distinctive Ringtone	CustomRing Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses 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.
Disable Call Waiting	Default is **No**.
Disable Call Waiting Tone	Default is **No**.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
No Key Entry Timeout	Default is **4 seconds**.
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.

|	
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 "(Re)Dial" key. If set to "No", this "#" key can be included as part of number.
Subscribe for MWI	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors.
Preferred Vocoder	The GXW–400x supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B. 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".
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Defines payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
G726-16 Payload type	Default value is 98. Range is from 96 to 127.
G726 - 24 Payload type	Default value is 99. Range is from 96 to 127.
G726 - 32 Payload type	Default value is 100. Range is from 96 to 127.
G726 - 40 Payload type	Default value is 103. Range is from 96 to 127.
G729E payload type	Default value is 102. Range is from 96 to 127.
VAD	Default is **No**.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
SLIC Setting	Dependent on standard phone type (and location)
Caller ID Scheme	• Bellcore/Telcordia • ETSI-DTMF • ETSI-FSK • SIN 227 - BT • NTT Japan

Polarity Reversal	Default is **No**. If set to "Yes", polarity will be reversed upon call establishment and termination.
Hook Flash timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.
Gain	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are **0dB** for both parameters. Loudest volume: +6dB Lowest volume: -6dB.
Call Progress/ Ring Tones	Configure ring or tone frequencies according to preference. By default tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds.

SAVING THE CONFIGURATION CHANGES

Once a change is made, press the "Update" button in the Configuration Menu. The following screen will confirm that the changes have been saved. To activate changes, reboot or power cycle the GXW–400x after changes are made.

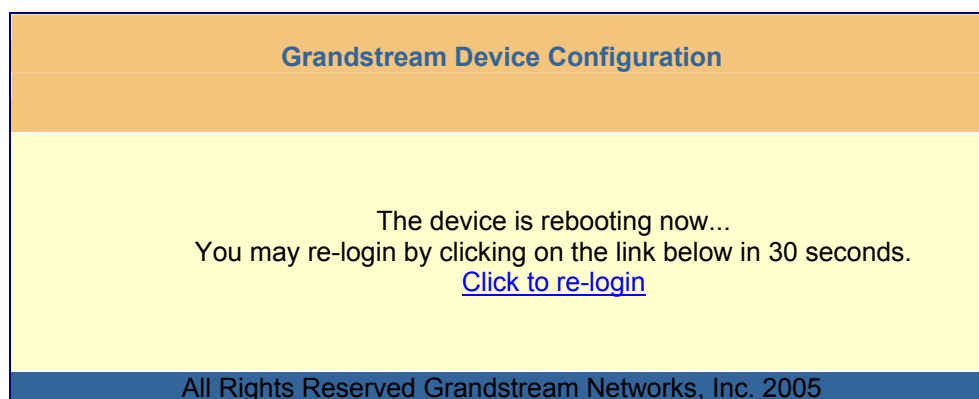FIGURE 4: SCREEN-SHOT OF SAVE CONFIGURATION PAGE
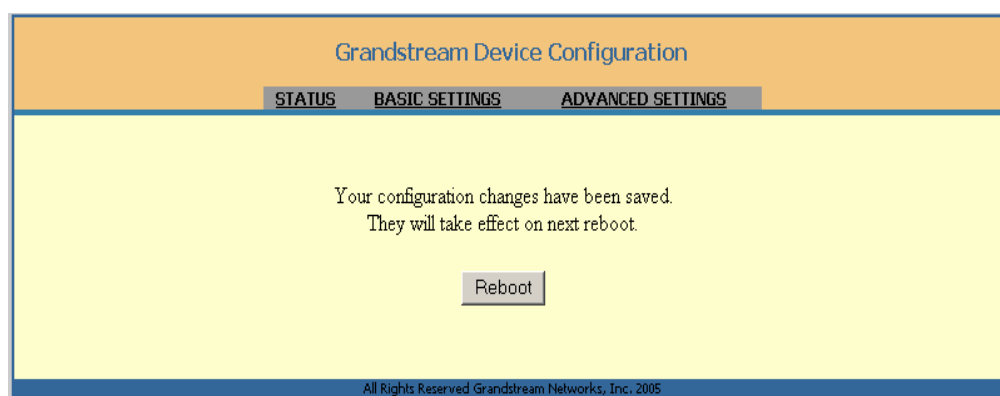
REBOOTING FROM REMOTE

The administrator can remotely reboot the unit by pressing the "Reboot" button at the bottom of the configuration menu. The user can re-login to the unit after waiting for about 30 seconds.

FIGURE 5: SCREEN-SHOT OF REBOOTING PAGE

FIRMWARE UPGRADE

Our latest official release can be downloaded from: http://www.grandstream.com/y-firmware.htm.

Firmware (or software) upgrades can be done either via TFTP or HTTP. The corresponding configuration settings are on the configuration page. End users should NOT touch the configuration settings that are useful for ITSPs. To upgrade your unit firmware, follow these steps:

1. Under Advanced Settings webpage, enter your TFTP or HTTP Server IP address (or FQDN) next to the "Firmware Upgrade: Upgrade Server" field.

2. Select via TFTP or HTTP accordingly.

3. If you plan to use Automatic Upgrade, set it to "Yes", otherwise No (this will make it check for upgrade every time you reboot).

UPGRADE THROUGH HTTP

To upgrade firmware via HTTP, the field *"Firmware Upgrade and Provisioning: Upgrade Via"* needs to be set to HTTP. The *"Firmware Server Path"* should be set to where the firmware files are located.

For example, the user can use the following URL in the Firmware Server Path:
firmware.mycompany.com: 6688/Grandstream/1.0.0.29 where firmware.mycompany.com is the FQDN of the HTTP server. It can also be in IP address format. ":6688" is the TCP port the HTTP server listening to, default http server listens to port 80. "/Grandstream/1.0.0.29" is the RELATIVE directory to the root dir on HTTP web server.

UPGRADE THROUGH TFTP

To upgrade firmware via TFTP, set the field *"Firmware Upgrade and Provisioning: Upgrade Via"* to TFTP. The TFTP server can be configured in either IP address format or FQDN.

To configure the TFTP server via the Web configuration interface, follow these five steps:

1. Open your browser to input the IP address of the GXW-400xv.
2. Enter the admin password to enter the configuration screen.
3. Enter the TFTP server address or URL in the "Firmware Server Path" field near the bottom of the configuration screen.
4. Once the "Firmware Server Path" is set, update the change by clicking the "Update" button.
5. Reboot or power cycle the unit.

If the configured updating server is found and a new code image is available, the GXW-400x will retrieve the new image files by downloading them into the GXW-400xv's SRAM. During this stage, the GXW-400xv's LED will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW-400xwill stop the TFTP process and simply boot using the existing code image in the flash.

A firmware upgrade may take 20 minutes over the Internet, or 20+ seconds if it is performed on a LAN. Grandstream recommends conducting firmware upgrades in a controlled LAN environment if possible.

No Local TFTP Server

For users who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Services section of Grandstream's Web site to obtain this TFTP server IP address. Alternatively, user can download and install a free TFTP or HTTP server in his LAN for a firmware upgrade.

A free Windows version TFTP server can be downloaded from:
http://support.solarwinds.net/updates/New-customerFree.cfm.

Downloading TFTP Server Directions:

1. Unzip the file and put all of the files under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW–400x in the same LAN segment.
3. Go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page.
5. Configure the Firmware Server Path with the IP address of the PC.
6. Update the change and reboot the unit.

You can also download the free HTTP server from http://httpd.apache.org or just use Microsoft IIS web.

RESTORE FACTORY DEFAULT SETTING

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.

FACTORY RESET

There are three (3) 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 <u>next to the power connection</u>.

3. Insert a pin in this hole, and press for about 7 seconds.

4. Take out the pin. All unit settings are restored to factory settings.

MAC Address

Reset default factory settings by changing 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:
   0-9: 0-9
   a. A: 22 (press the "2" key twice, "A" will show on the LCD)
   b. B: 222
   c. C: 2222
   d. D: 33 (press the "3" key twice, "D" will show on the LCD)
   e. E: 333
   f. F: 3333

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

NOTE: If there are digits like "22" in the MAC, you need to type "2" then press "->" right arrow key to move the cursor or wait for 4 seconds to continue to key in another "2".

IVR Command

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

1. Dial "***" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the MAC address (on bottom of unit)
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

GLOSSARY OF TERMS

ADSL Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC Automatic Gain Control is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

ATA Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

CODEC Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG Comfort Noise Generator, generate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

DATAGRAM A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while sub-sampling.

DECT Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multi-carrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multi-frame (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS Short for *D*omain *N*ame *S*ystem (or *S*ervice or *S*erver), an Internet service that translates *domain names* into IP addresses

DID Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

DSP Digital Signal Processor. A specialized CPU used for digital signal processing. Grandstream products all have DSP chips built inside.

DTMF Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #).

FQDN Fully Qualified Domain Name. A FQDN consists of a host and domain name, including top-level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, Grandstream is the second-level domain, and and.com is the top level domain.

FXS Foreign eXchange Office. An FXS device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.
  • An FXS interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.
  • FXS is complimentary to FXS (and the PSTN).

FXS Foreign eXchange Station. An FXS device has hardware to generate the ring signal to the FXS extension (usually an analog phone).
  • An FXS device will allow any FXS device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.
  • The FXS Interface connects to FXS devices (by an FXS interface, of course).

DHCP The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network. There are **two types** of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323 A suite of standards for multimedia conferences on traditional packet-switched networks.

HTTP Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP Internet Protocol. A packet-based protocol for delivering data across networks**.**

IP-PBX IP-based Private Branch Exchange

IP Telephony (Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

MTU A Maximum Transmission Unit (MTU) is the largest size packet or frame, specified in octets (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT Network Address Translation

NTP Network Time Protocol, a protocol to exchange and synchronize time over networks The port used is UDP 123 Grandstream products using NTP to get time from Internet

OBP/SBC Outbound Proxy or another name Session Border Controller. A device used in VoIP networks. OBP/SBCs are put into the signaling and media path between calling and called Caller. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called Caller. The effect of this behavior is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with firewalls to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with internet connections using NAT.

PPPoE Point-to-Point Protocol over Ethernet is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN Public Switched Telephone Network. The phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP Real-time Transport Control Protocol, defined in RFC 3550, a sister protocol of the Real-time Transport Protocol (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889

SDP Session Description Protocol is a format for describing streaming media initialization parameters. It has been published by the IETF as RFC 2327.

SIP Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323. All Grandstream products are SIP based

STUN Simple Traversal of UDP over NATs is a network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489. STUN will usually work well with non-symmetric NAT routers.

TCP Transmission Control Protocol is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create *connections* to one another, over which they can exchange data or packets. The protocol guarantees reliable and in-order delivery of sender to receiver data.

TFTP Trivial File Transfer Protocol, is a very simple file transfer protocol, with the functionality of a very basic form of FTP; It uses UDP (port 69) as its transport protocol.

UDP User Datagram Protocol (UDP) is one of the core protocols of the <u>Internet protocol suite</u>. Using UDP, programs on networked computers can send short messages known as *<u>datagrams</u>* to one another. UDP does not provide the reliability and ordering guarantees that <u>TCP</u> does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD Voice Activity Detection or Voice Activity Detector is an algorithm used in <u>speech processing</u> wherein, the presence or absence of human speech is detected from the audio samples.

VLAN A virtual <u>LAN</u>, known as a VLAN, is a logically-independent <u>network</u>. Several VLANs can co-exist on a single physical <u>switch</u>. It is usually refer to the <u>IEEE 802.1Q</u> tagging protocol.

VoIP Voice over the Internet. VoIP encompasses many protocols. All the protocols do some form of signaling of call capabilities and transport of voice data from one point to another. e.g.: SIP, H.323, etc.