

User Manual

GXV-3000

Innovative IP Video Phone

For Firmware Version 1.0.0.27



Grandstream Networks, Inc.

www.grandstream.com

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1 Welcome

Thank you for purchasing Grandstream award-winning GXV-3000 Video IP Phone. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream GXV-3000 SIP IP Video Phone is a next generation advanced IP videophone based on SIP and H.264 standard. Built upon Grandstream's innovative technology, the GXV-3000 IP Video telephone offers a rich set of functionality, ease of use, superb sound and video quality, stylish exterior design and highly attractive price. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

This document is subject to changes without notice. The latest electronic version of this user manual is available for download from the following location:

http://www.grandstream.com/user_manuals/GXV-3000.pdf

Award Winning Products



2 Installation

2.1 What is Included in the Package

The GXV-3000 phone package contains:

- 1) One GXV-3000 Main Case
- 2) One Handset
- 3) One Phone Cord
- 4) One Universal Power Adaptor
- 5) One Ethernet Cable
- 6) Two Wall Mount Sustaining Brackets

2.2 Connecting Your Phone

Following is a backside picture of GXV-3000, each connection port is labeled with the name in the following table:



Table 2-1: The connectors on the GXV-3000 phone:

USB	USB Interface for external USB device like flash to store captured image, ring tones and phone address books, etc.
LAN	10/100 RJ-45 port for uplink to Ethernet; or WAN port when configured in Router mode.
PC	10/100 RJ-45 port for connecting PC in switch mode or LAN port in router mode

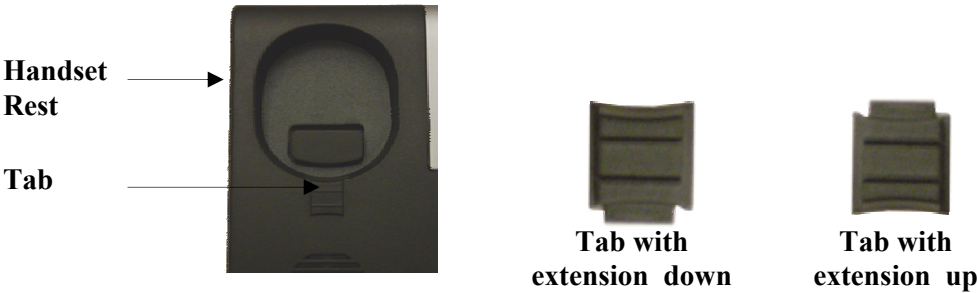
Power	12V DC power port
RCA Output	RCA Output Interface to external display like TV set
Headset	2.5mm Headset port

2.3 Wall Mount

GXV-3000 can be wall mounted. The two wall mount sustaining brackets can be attached to the bottom of the main body:



User can simply place the device against the wall with two holes to the fixed hanger and the two sustaining brackets balance the bottom to position the phone on the wall.



User will need to pull out the tab (extension downward) from handset cradle on the top of the handset rest, and rotate the tab and plug it back into the slot with the extension up for handset holding.

2.4 Safety Compliances

The GXV-3000 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only be operated with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors are not covered by the manufacturer's warranty.

2.5 Warranty

Grandstream has a reseller agreement with our reseller customer. End user 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 before you return the product.

Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the GXV-3000 and will void the manufacturer warranty.

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.

3 Product Overview

GXV-3000 IP Video Phone is designed to be used in general household or office. The following photo illustrates the appearance of a GXV-3000 IP Video phone.

Front View



Back View












3.1 Key Features

Grandstream GXV-3000 IP Video Phone is a next generation advanced IP video telephone based on industry's open standard SIP (Session Initiation Protocol) and H.264. Built upon Grandstream's innovative technology, GXV-3000 IP Video Phone features superb audio and video quality, rich functionalities, stylish exterior design and highly attractive price.

Software Feature:

- ✚ Support SIP 2.0 , TCP/UDP/IP , RTP/RTCP , HTTP/HTTPS , ICMP , ARP/RARP , DNS, DHCP (both client and server), NTP, PPPoE, TFTP, Telnet, TLS (pending), etc.
- ✚ Support dual 10M/100M auto-sensing Ethernet ports configurable to operate under either switch or router/NAT mode (pending).
- ✚ Support Quick IP Call Mode (partially simulating PBX in a LAN environment w/o SIP server)
- ✚ Powerful video DSP with advanced adaptive jitter control and packet loss concealment technology to ensure superb audio and video quality
- ✚ Support advanced H.264 base line real-time video codec (at CIF or QVGA resolution and up to 30 frames/second) to ensure highest quality video delivery at 32kbps – 1Mbps bandwidth level (bit rate/frame rate configurable to reach best audio/video quality in available bandwidth)
- ✚ Support various audio codecs including G.711 A/U law (PCMA/PCMU), G.723.1, G.729A/B, GSM, G.726 (pending) and iLBC (pending) with dynamic negotiation of codec type and packet time during call setup
- ✚ Support popular voice features including 3 line indicators (each of which can be configured using independent SIP account), Caller ID/Name Display/Block, Hold, Call Waiting, Call Transfer, Call Forward (no answer, busy, unconditional), Full Duplex Hands-Free Speaker Phone with advanced Acoustic Echo Cancellation (AEC/G.167), 3-way Conference, Do-not-Disturb, Incoming Call Rejection, various DTMF options (in audio, RFC2833, SIP INFO), downloadable music ring tone, etc.
- ✚ Support advanced video features including a high quality 5.6-inch TFT color LCD (allowing 2-dimensional view angle adjustment including 180 degrees vertical rotation and nearly 300 degrees horizontal rotation), advanced VGA resolution CMOS camera sensor (view angle adjustable), anti-flickering, auto focus and auto exposure, 2X analog and 4X digital Zoom, PIP (Picture-in-Picture), audio mute and camera block (for privacy), call log, phone/address book,

configurable screen-saver pictures, still picture capture/store/send (VGA resolution), visual voice message indicator, intuitive graphic user interface enabled by 5 navigation buttons, etc.

-  Support 2 USB (2.0) host ports, 1 audio and 1 video output jack (capable of outputting video to an external TV simultaneously), headset jack (2.5mm)
-  Support silence suppression and VAD, AGC and acoustic echo cancellation (G.167)
-  Support standard encryption and authentication using DIGEST (MD5 and MD5-sess) and AES
-  Support secure signaling (SIP over TLS, pending) and secure voice/video communication (SRTP) (pending)
-  Support layer-2 (802.1Q VLAN, 802.1p) and layer-3 (DiffServ, ToS) QoS
-  Support automated NAT traversal without manual manipulation of firewall/NAT
-  Support remote automated and secure provisioning and software upgrade through firewall/NAT to enable “zero configuration” and “plug-and-dial” for end users
-  Support remote device monitoring and events reporting using Syslog
-  Support device configuration via LCD, Web browser or central secure configuration file

3.2 Hardware Specification

The table below describes the hardware specification of GXV-3000:

Model	GXV-3000
Ethernet Port	Dual 10M/100M auto-sensing Ethernet ports Switch or router/NAT mode configurable
LCD	5.6 inch TFT color LCD
Camera	Advanced CMOS sensor (VGA resolution), 330K Pixels
Auxiliary Port	RCA style stereo audio & composite video jack, 1 headset jack, 2 USB 2.0 ports
Exterior	Black or Silver ABS plastic, 30 buttons (6 of which are LED buttons)
Headset Jack	2.5mm Headset Port
Universal Power Supply	100-240V input, +12VDC/1.2A output US/Euro/UK/Japan/Australian style available
Dimension	6.5cm x 18cm x 16cm
Weight	1.2kg
Operating Temperature	0-40 °C
Humidity	10-95% non-condensing
Compliance	FCC/CE/C-Tick (pending)




4 Using GXV-3000 IP Video Phone




4.1 Getting Familiar with LCD

GXV-3000 IP Video phone has a 5.6 inch TFT color LCD. Here is the sample LCD display showing the phone is registered to VoIP service providers' SIP server/proxy with one missed call icon.



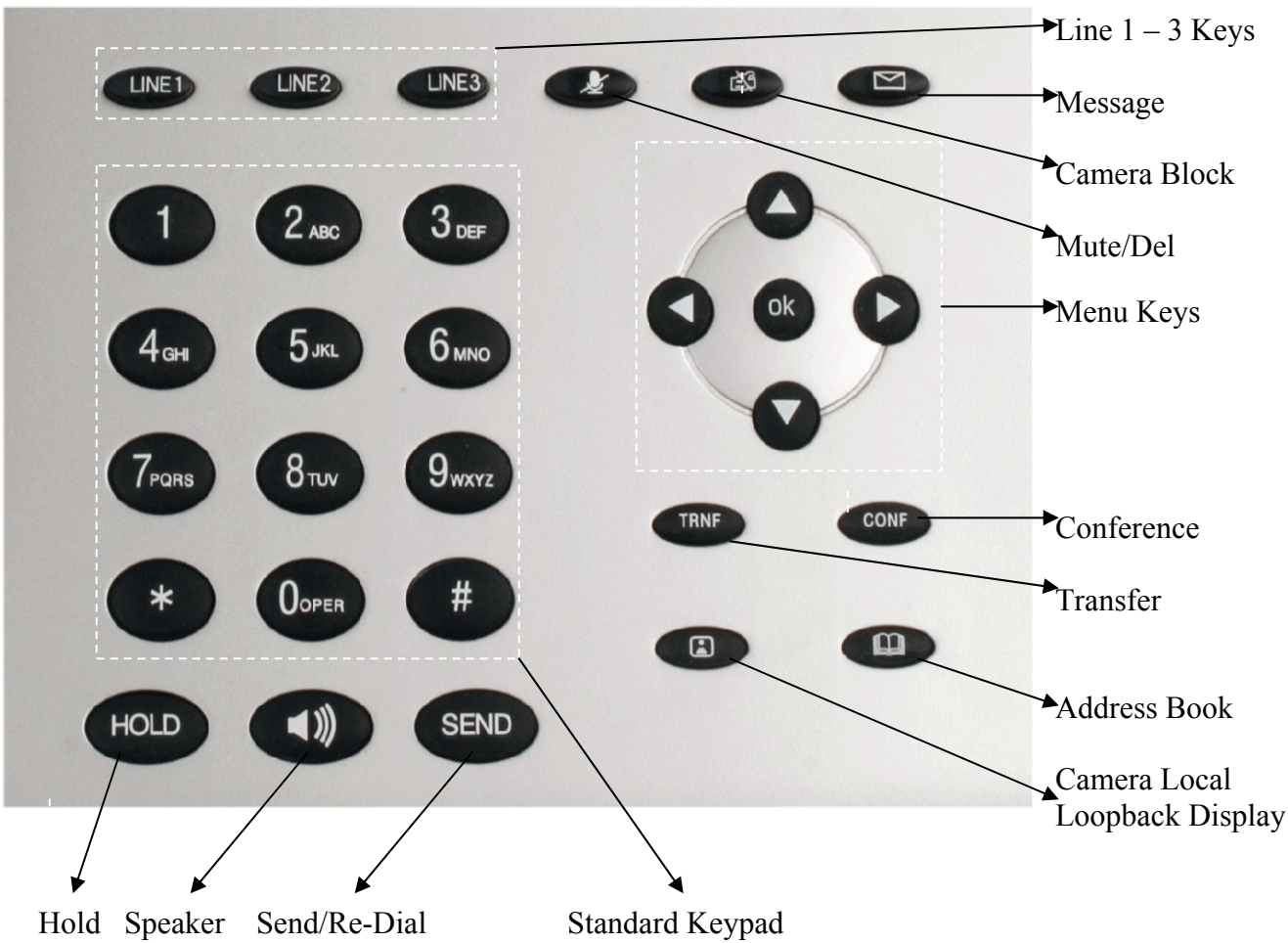
The phone has a screen saver. When it is configured, the phone will show the screen saver or turn off the LCD just like normal laptop computer screen.

Icon	LCD Icon Definitions
	Network Status Icon: RED and FLASH in the case of Ethernet link failure RED if IP address or SIP server is not found ON if IP address and SIP server are located
	Phone Status Icon: OFF when the handset is on-hook ON when the handset is off-hook
	Speaker Phone Status Icon: FLASH when phone rings or a call is pending OFF when the speakerphone is off ON when the speakerphone is on






	Handset, Speakerphone and Ring Volume Icon: 0-7 scales to adjust handset / speakerphone / ring volume
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
	Time Icon: AM for the morning PM for the afternoon


4.2 Getting Familiar with Keypad

Here is the key assignment in the phone’s keypad:



GXV-3000 IP Video phone has 30 key buttons:

Key Button	Key Button Definitions
LINE1-LINE3	3 Line keys with LED, can be configured to different SIP profile
UP ↑	Next “Menu Item” when phone is in keypad configuration mode Or increase handset/speakerphone volume when phone is ACTIVE Or increase ring volume when phone is in IDLE mode
DOWN ↓	Previous “Menu Item” when phone is in keypad configure mode Or decrease handset/speakerphone volume when phone is ACTIVE Or decrease ring volume when is in IDLE mode
LEFT ←	Shift cursor to left
RIGHT →	Shift cursor to right
OK	Enter Keypad Configuration “MENU” mode when phone is in IDLE mode. It is also the ENTER key once entering Keypad Configuration.
TRNF	TRANSFER key: Transfer an ACTIVE call to another number
CONF	Bring Calling/Called party into conference
 MESSAGE	Enter to retrieve (video) voice mails or other messages
 CAMERA BLOCK	Enable/Disable Block Camera for private, just doing normal voice/audio call without sending video. When this turned on, the phone is just function as a normal IP phone
 MUTE/DEL	Enable/Disable to Mute an ACTIVE call; or function as Delete a key entry, call log, etc; or shortcut to Activate/Deactivate DND when idle; or Reject an incoming call when ringing
 CAMERA LOCAL LOOPBACK DISPLAY	Enable/Disable camera local loopback display
 ADDRESS BOOK	Video Phone Address Book (Not fully ready yet, will implement downloadable address book using XML in future f/w)

 SPEAKER	Enable/Disable hands-free speaker mode
HOLD	Temporarily hold an ACTIVE call
SEND	Dial a new number or Redial the last number dialed. After entering the phone number, pressing this key would force a call to go out immediately before “no key entry timeout” value expires
0 - 9, *, #	12 standard Digit, * and # keys are usually used to make phone calls

4.3 Making and Answering Phone Calls

4.3.1 Handset, Speakerphone and Headset Mode

Handset can be switched between either Speaker or Headset. However, whenever the Headset is plugged in, Speaker will be switched to Headset.

To Switch between Handset and Speaker/Headset, simply press the Hook Flash in the Handset cradle or press the SPEAKER button in the phone.

4.3.2 Multiple SIP Accounts and Lines

GXV-3000 can support up to 3 independent SIP accounts. Each account is capable of independent SIP server, user and NAT settings among others. Each of the 3 LINE buttons (LINE1-LINE3) is “virtually” mapped to each SIP account. In off hook state, when user chooses an idle line, the name of the account (as configured in the web interface) will be displayed in the LCD while a dial tone is being played out. For example, if the 3 SIP accounts are named FWD (FreeWorldDialup), BroadVoice and Asterisk PBX respectively and they are all active and registered. When LINE1 is pressed, user will hear dial tone and see “FWD”. When LINE2 is pressed, user will hear dial tone and see “BroadVoice”. When LINE3 is pressed, user will hear dial tone and see “Asterisk PBX”.

For outgoing calls, GXV-3000 will pick up the LINE pressed, which the LED will be lit up in solid red color. User can switch the dialing account before dialing any digits by pressing the same LINE button one or more times. If user continues to press one LINE, the selected account will circulate among the registered accounts. For example, when LINE1 is pressed, LCD displays “FWD”. If LINE1 is pressed twice, LCD displays “BroadVoice” and the subsequent call will be made through SIP account 2.

For incoming calls, if an account is configured and registered, all incoming calls for that account will attempt to use its corresponding LINE if it is not in use. When the “virtually” mapped line is in use, GXV-3000 will flash the next available LINE (from Left to Right) in red color.

A LINE is defined as “ACTIVE” when it is making or receiving a call, and its corresponding LINE LED is lit up in solid RED.

4.3.3 Making Calls

There are four ways to make phone calls:

1. Make Handset/SPEAKER/Headset off hook, or press the available LINE key to select a SIP account, the corresponding LINE LED will light up in solid red. Enter the phone numbers and press the SEND key.
2. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up in solid red. Press the SEND button to redial the last number called.
3. Press the OK to bring up the Main Menu, select Phone Book, browse phone book to the person you want to dial, press OK to select and OK again when Dial is selected. The call will dial out in SPEAKER mode. (Right now this only applies to primary a/c which is LINE 1)
4. Paging/Intercom: This is ONLY valid if the SERVER/PBX support this feature and both phone and PBX (for example, Asterisk) are configured correctly:
Make Handset/SPEAKER/Headset off hook, select the LINE key to related feature available account, then press OK key again to see LCD showing: LINEx: PAGE USING. Now input the phone number you want to Page/Intercom and press SEND key.

NOTE:

- Once off hook and key pressed, the dialed number will be displayed on the LCD and the corresponding DTMF tone will be played out.
- If the “SEND” button is not pressed after the phone number pressed, the phone (by default) will wait for 4 seconds (no key entry timeout) before sending all digits out and initiating the call.

4.3.4 Making Calls using IP Address

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if

- Both phones have public IP addresses, or
- Both phones are on a same LAN/VPN using private or public IP addresses, or
- Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

To make a direct IP calling, press “OK” button to bring up MAIN MENU, then select “Direct IP Call” in the MENU and press OK to select, then key in the 12-digit target IP address, press OK key again to call out.

Examples:

If the target IP address is 192.168.1.60 and port is 5062, like 192.168.1.60:5062, the key input to the Menu UI will be:

192*168*1*60#5062

followed by OK key to dial out, where “*” key represent “.” and “#” key represent “:”

- **Quick IP Call Mode:**

This model has the ability to dial an IP address under the same LAN/VPN segment by simply pressing the last octet in the IP address. This will allow user partially simulate a PBX functions using CMSA/CD without a SIP server. When doing this, controlled static IP usage are recommended.

In the “Advanced Settings” page there is an option "Use Quick IP-call mode", by default it is set to No. When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask.

#XX or #X are also valid so leading 0 is not required (but OK).

eg.

192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or #

192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or #

192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #

192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3

NOTE:

- If you have a SIP Server configured, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.
- When doing Direct IP calling, make sure the “Use Random Port” is configured as “NO”. Otherwise it will not work unless the phone is configured to listen to the special port you specified and also that port need to be inputted in the dialing digits.

4.3.5 Receiving Calls

There are three states when GXV-3000 receives a call:

1. When receiving an initial call. Besides ringing with selected Ring Tone, the corresponding account LINE will flash in red, taking Handset/SPEAKER/Headset off hook will enable user to hear the calling party in Handset/SPEAKER/Headset.
2. When receiving second or more incoming calls, besides playing stutter Call Waiting tone, GXV-3000 will pick up the corresponding account LINE or the next available LINE as described in section 4.3.2.
3. When the related configuration is enabled for Paging/Intercom and the PBX (or Server) also supports this feature, the phone will beep once and automatically establish the call via SPEAKER.

4.3.6 Call Hold

While in conversation, pressing the “HOLD” button will put the other party on hold. User can resume the conversation by pressing the corresponding blinking LINE. User will also automatically put the current line on “HOLD” by pressing another available LINE for making or receiving other phone calls.

4.3.7 Call Waiting and Switch between Calls

GXV-3000 can switch to another line for making or answering calls and automatically put an ACTIVE call on Hold.

When receiving second or more incoming calls, besides playing a stutter Call Waiting tone, GXV-3000 will pick up the corresponding account or the next available LINE as described in section 4.3.2.

4.3.8 Call Transfer (Pending: Only work in Audio mode)

GXV-3000 supports both BLIND and ATTENDED Transfer:

1. Blind Transfer: When a LINE is “ACTIVE”, user will get a dial tone by pressing the “TRNF” button, then dial the number and press the “SEND” button, this will transfer the other party in the corresponding LINE to the dialed number.
2. Attended Transfer: When in conversation with an “ACTIVE” LINE as defined in section 4.3.2, user shall press “TRNF” button, then press the intended blinking LINE that is on “HOLD”.

If there is no LINE on HOLD, user will need to make a call and thus automatically puts the current ACTIVE LINE on HOLD.

NOTE:

- Transferring calls across SIP domains needs to be supported by SIP service providers.

4.3.9 3-Way Conferencing (Pending: Only work in Audio mode)

GXV-3000 supports 3-way conferencing. With one LINE ACTIVE and another LINE on HOLD, press the CONF button then the LINE that is on HOLD will join the three parties together in a conference.

If after pressing the “CONF” button, a user decides not to conference anyone, user can cancel it and resume the conversation by pressing CONF or the original LINE button.

If the conference holder wishes to end a conference, simply press HOLD, which breaks the conference and places both parties on hold. User can then talk to each individual party by selecting the corresponding LINE.

4.3.10 Checking Message and Message Waiting Indication

When GXV-3000 is on-hook, pressing the MSG button will trigger the phone to call the VoiceMail Server (VMS) configured for the primary account. If a line/account is selected first, it dials the VMS configured for that account.

The MWI (Message Waiting Indicator) LED will flash in red color in three quarters of a second when voicemail server sends message waiting information to GXV-3000.

NOTE:

- This feature requires the phone to be configured correctly in the “VoiceMail User ID” field the VM portal access number, also requires the Server to support this.

4.3.11 Mute / Delete

When in conversation with an ACTIVE LINE, pressing “MUTE/DEL” will mute the conversation, which means you can hear the other party but the other party cannot hear you. Pressing the button again will resume the conversation. When pressed, the red icon (muted microphone) will blink in the LCD to remind call is muted.

When dialing a number or during the Key Pad “MAIN MENU” configuration mode, press “MUTE/DEL” will delete the last entered digit.

When phone is idle, press the Mute/Delete key will function as a shortcut to Enable/Disable the DND (Do Not Disturb). When enabled, the red icon (Do Not Disturb) will blink in the LCD to remind user the DND is activated.

When there is an incoming call, press the Mute/Delete key while phone is ringing will REJECT the incoming call. The server will tell caller the user is busy and/or forward the call to Voice Mail if server support this feature and configured properly.

4.3.12 CAMERA BLOCK

User can press CAMERA BLOCK key to Disable/Enable the video sending out when in video call. If no video call wanted (privacy), user can press this button and make Audio Only call. When CAMERA BLOCK key is pressed, a red blocked camera will blink in the LCD to remind that the call is not in video mode and the camera is blocked.

4.4 Call Features

GXV-3000 series phone supports a list of call features: Caller ID Block (or Anonymous Call), Disable/Enable Call Waiting, Call Forward on Busy, Delay, or Unconditional, etc.

Following table shows the call features of GXV-3000 series phone.

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward To use this feature, dial “*72” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial “*90” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*91	Cancel Busy Call Forward To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
*92	Delayed Call Forward To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial “*93” and get the dial tone, then hang up.

5 Configuration Guide

5.1 Configuration with Keypad

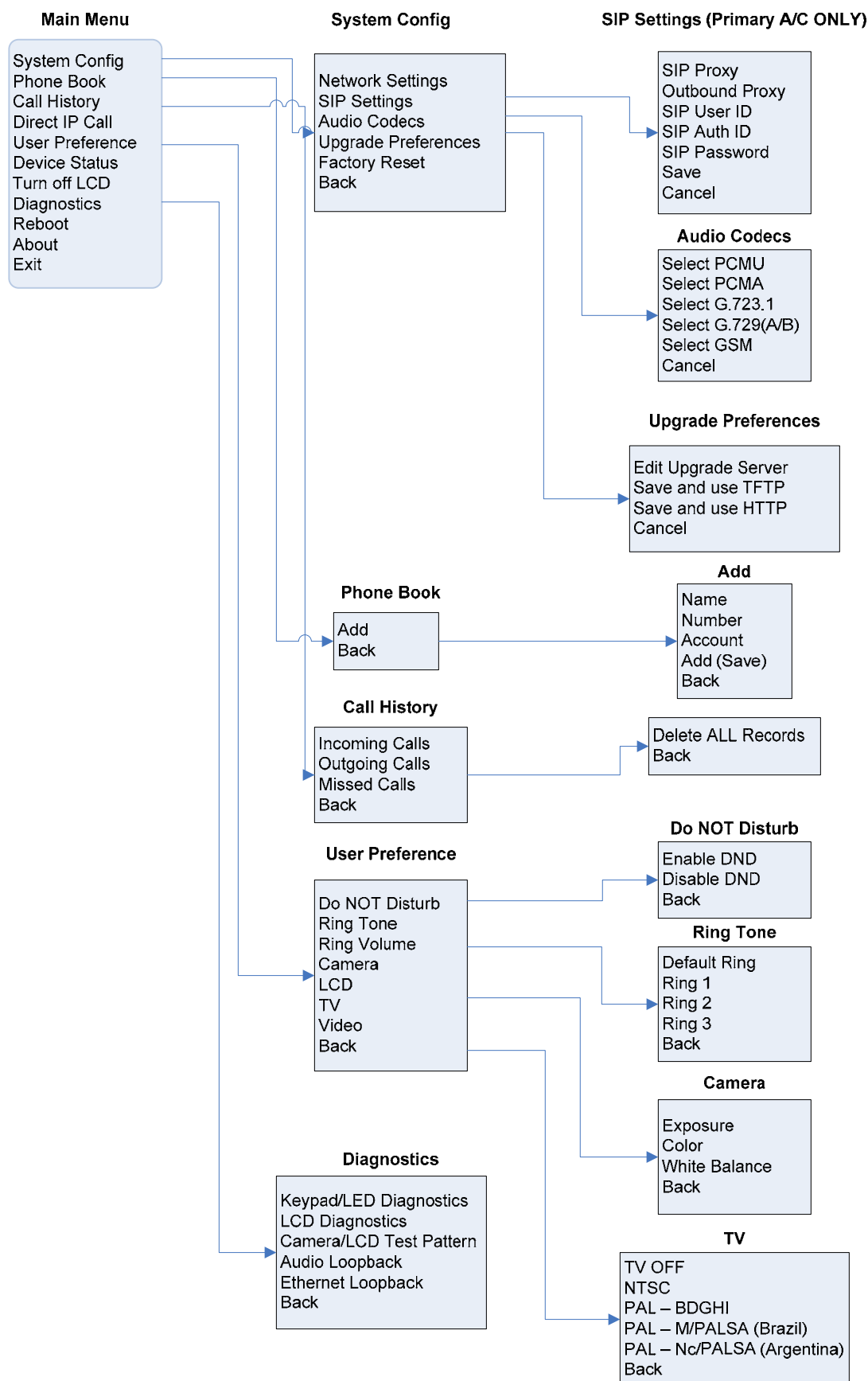
When the phone is on hook, press the OK button to enter MENU mode. When the phone goes off hook or a call comes in, the phone automatically exits the MENU state and prepares for the call. It also exits the MENU state if left idle for 20 seconds.

Here are the Menu options supported:



Here is the flow chart of keypad Menu configuration. During operation of the keypad, OK key is function as select or confirm key (similar to Enter in PC), UP and DOWN arrow keys are functioning as browsing (by highlight the item selected), LEFT and RIGHT arrow keys are functioning as input selection, MUTE/DEL key is functioning as delete if input is wrong.

Table 5-1: Flow Chart of Key Pad Configuration Menu:



5.2 Configuration with Web Browser

GXV-3000 series IP phone has an embedded Web server that will respond to HTTP/HTTPS GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through a Web browser such as Microsoft's IE or Mozilla Firefox.

5.2.1 Access the Web Configuration Menu

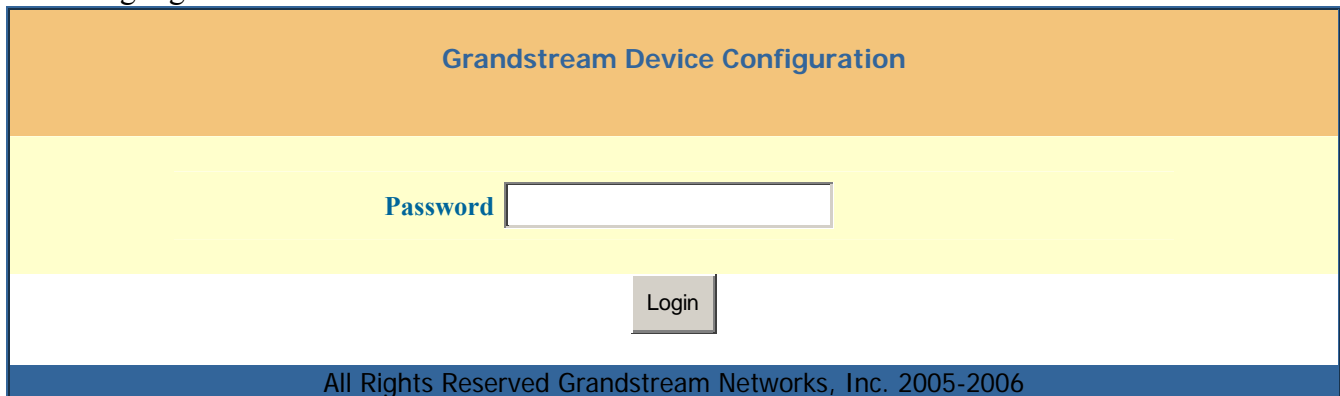
The IP Phone Web Configuration Menu can be accessed by the following URI:

<http://Phone-IP-Address>

where the *Phone-IP-Address* is the IP address of the phone, which showed in the LCD screen.

5.2.2 End User Configuration

Once this HTTP request is entered and sent from a Web browser, the GXV-3000 will respond with the following login screen:



Grandstream Device Configuration

Password

Login

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The password is case sensitive with maximum length of 25 characters and the factory default password for End User is "123", for Administrator is "admin". Only administrator has the privilege to access the Advanced Setting and Account information. User trying to access without privilege will get an error.

After a correct password is entered in the login screen, the embedded Web server inside the GXV-3000 will respond with the Configuration page, which is explained in details below. Following is screen shot of BASIC SETTINGS page when user using End User privilege login will see:

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
Web Access: <input checked="" type="radio"/> HTTP <input type="radio"/> HTTPS	
Web Port: <input type="text" value="80"/> (default for HTTP is 80 and HTTPS is 443)	
End User Password: <input type="text"/> (purposely not displayed for security protection)	
IP Address: <input checked="" type="radio"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE first if PPPoE setting is non-blank)	
<div> <div>DHCP hostname:</div> <div><input type="text"/></div> </div>	
<div> <div>DHCP domain:</div> <div><input type="text"/></div> </div>	
<div> <div>DHCP vendor class ID:</div> <div><input type="text"/></div> </div>	
<div> <div>PPPoE account ID:</div> <div><input type="text"/></div> </div>	
<div> <div>PPPoE password:</div> <div><input type="text"/></div> </div>	
<div> <div>PPPoE service name:</div> <div><input type="text"/></div> </div>	
Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>	
<input checked="" type="radio"/> statically configured as:	
<div> <div>IP Address:</div> <div> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> </div> </div>	
<div> <div>Subnet Mask:</div> <div> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> </div> </div>	
<div> <div>Default Router:</div> <div> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> </div> </div>	
<div> <div>DNS Server 1:</div> <div> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> </div> </div>	
<div> <div>DNS Server 2:</div> <div> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> </div> </div>	
Time Zone: <input type="text" value="GMT-5:00 (US Eastern Time, New York)"/>	
Daylight Savings Time: <input type="radio"/> No <input checked="" type="radio"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)	
Time Display Format: <input checked="" type="radio"/> 12 HOUR <input type="radio"/> 24 HOUR	
Date Display Format: <input checked="" type="radio"/> Year-Month-Day <input type="radio"/> Month-Day-Year <input type="radio"/> Day-Month-Year	
Display Clock instead of Date: <input type="radio"/> No <input checked="" type="radio"/> Yes	
LCD Screen Saver Interval: <input type="text" value="1800"/> (in seconds. 0 means screen saver is off)	

LCD Auto Power Off Interval:	<input type="text" value="0"/>	(in seconds. 0 means LCD will be always on)		
LCD Brightness:	<input type="text" value="196"/>	(from 0-255, default is 128)		
LCD Contrast:	<input type="text" value="132"/>	(from 0-255, default is 128)		
LCD Chroma Saturation:	<input type="text" value="132"/>	(from 0-255, default is 128)		
LCD Background Color:	Red (0-255) <input type="text" value="0"/>	Green (0-255) <input type="text" value="0"/>	Blue (0-255) <input type="text" value="0"/>	
OSD Text Color:	<input type="text" value="Blue"/>			
Camera zoom mode:	<input type="text" value="1.0:1 (wide)"/>			
Camera Exposure:	<input type="text" value="Auto"/>			
Camera Color Mode:	<input checked="" type="radio"/> Color	<input type="radio"/> Monochrome		
Camera White Balance:	<input checked="" type="radio"/> Auto	<input type="radio"/> Fixed		
Camera Lens Correction:	<input type="radio"/> No	<input checked="" type="radio"/> Yes		
TV Output:	<input type="text" value="Off"/>			
<input type="button" value="Update"/>				
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Web Access	Select HTTP or secure HTTPS protocol for Web Access
Web Port	By default, HTTP uses port 80 and HTTPS uses port 443. This field is for customizable web port.
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
IP Address	<p>There are two modes under which the GXV-3000 can operate:</p> <ul style="list-style-type: none"> • If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXV-3000 will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. The DHCP option is reserved for NAT router mode (not ready yet) • To use the PPPoE feature the PPPoE account settings need to be set. The GXV-3000 will attempt to establish a PPPoE session if any of the PPPoE fields is set. • If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are set to zero by default.

Time Zone	This parameter controls how the date/time is displayed according to the specified time zone.
Daylight Savings Time	This parameter controls whether the time will be displayed in daylight savings time or not. If set to “Yes”, then the displayed time will be 1 hour ahead of normal time.
Time Display Format	LCD time display in 12 hour or 24 hour format
Date Display Format	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year
Display Clock instead of Date	LCD displays clock if set to “Yes”, it will show clock in the LCD. Default is No.
LCD Screen Saver Interval	Default value is 60 seconds, thus after 60 seconds of idle time the LCD Screen Saver will be enabled. Currently the screen saver file limitation is 20 images or 320kb whichever limit is reached first. Screen saver picture can be changed the same way as customized ring tone.
LCD Auto Power Off Interval	Default value is 300 seconds, thus after 300 seconds of idle time the LCD will automatically Power Off.
LCD Brightness	Default value is 128
LCD Contrast	Default value is 128
LCD Chroma Saturation	Default value is 128
LCD Background Color	Select desired LCD Background color from the drop down box. Initial all is 0, can be adjusted to the color designed
OSD Text Color	Select desired On Screen Display Text color from drop down box. Default is blue. The color can distinguish OSD from video background
Camera Zoom Mode	Select desired Camera zoom mode (Tele, optical, digital). Can adjust local video zoom either in mirror mode or during the call on the fly by press the Left or Right Arrow Key
Camera Exposure	Select desired Camera Exposure mode based on surroundings.
Camera Color Mode	Select desired Camera Color Mode (Color or Monochromatic)
Camera White Balance	Select desired Camera White Balance to be automatic or Fixed.

Camera Lens Correction	Correct Camera Lens noise. Default is No. Recommend to set to YES to reduce camera lens noise and improve video quality.
TV Output	Select desired TV Output type (PAL or NTSC).

In addition to the Basic Settings configuration page, end user also has access to the device Status page. The following is a screen shot of the device Status page. Details are explained next.

Grandstream Device Configuration	
STATUS	BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3
Hardware Revision:	Main-- 1.0 Rev A Interface-- 1.0 Rev A
MAC Address:	00.0B.82.08.77.5A
IP Address:	192.168.10.135
Product Model:	GXV3000
Software Version:	Application-- 1.0.0.27 Loader-- 1.0.0.2 Boot-- 1.0.0.0
System Up Time:	2 day(s) 6 hour(s) 35 minute(s)
Registered:	Account 1: Yes Account 2: No Account 3: No
PPPoE Link Up:	disabled
NAT:	detected NAT type is full cone
NAT Mapped IP:	72.72.12.143
Total Inbound Calls:	2
Total Outbound Calls:	3
Total Missed Calls:	0
Total SIP Message Sent:	1085
Total SIP Message Received:	1077
Total RTP Packet Sent:	77523
Total RTP Packet Received:	73774
Total RTP Packet Loss:	270
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Hardware Revision	Hardware version number: Main Board, Interface Board
MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows IP address of GXV-3000

Product Model	This field contains the product model information.
Software Version	<ul style="list-style-type: none"> • Program: This is the main software (firmware) release number, always used for identify the whole software (firmware) system of the phone. • Loader: Driver loader code version number. • Boot: Booting code version number
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the accounts are registered to the related SIP server(s). GXV-3000 can support three different SIP profile.
PPPoE Link Up	This field shows whether the PPPoE connection is up if connected to DSL modem.
Detected NAT Type	This field shows what kind NAT router the GXV-3000 is connected to. It is based on STUN protocol resolution result.

5.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 5.2.1 to get to the following login page. The password is case sensitive with a maximum length of 25 characters and the factory default password for Advanced User is “admin”.

Grandstream Device Configuration

Password

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Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration. Following is a screen shot of the advanced configuration page:

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
	Admin Password: <input type="text"/> (purposely not displayed for security protection)
	G723 rate: <input checked="" type="checkbox"/> 6.3kbps encoding rate <input type="checkbox"/> 5.3kbps encoding rate
	Silence Suppression: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
	Voice Frames per TX: <input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
	Video frame rate: <input type="text" value="15 frames/second"/>
	Video bit rate (kbps): <input type="text" value="128 kbps"/>
	Video Rate Control: <input checked="" type="checkbox"/> Frame <input type="checkbox"/> TMN5 (better for bit rate >= 384)
	Video Frame Skipping: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
	Video Post Processing: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
	Video fit to screen: <input checked="" type="checkbox"/> Scaling <input type="checkbox"/> Cropping
	Layer 3 QoS: <input type="text" value="48"/> (Diff-Serv or Precedence value)
	Layer 2 QoS: 802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)
	No Key Entry Timeout: <input type="text" value="4"/> (in seconds, default is 4 seconds)
	Use # as Dial Key: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
	local RTP port: <input type="text" value="5004"/> (1024-65535, default 5004)
	Use random port: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
	keep-alive interval: <input type="text" value="20"/> (in seconds, default 20 seconds)
	Use NAT IP <input type="text"/> (if specified, this will be used in SIP/SDP message)
	STUN server: <input type="text" value="stun.fw.dnet.net:3478"/> (URI or IP:port)
	Firmware Upgrade: Upgrade Server: <input type="text" value="fm.grandstream.com/gs"/> Via <input checked="" type="checkbox"/> TFTP <input type="checkbox"/> HTTP Automatic Upgrade: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days)
	DTMF Payload Type: <input type="text" value="101"/>
	Syslog Server: <input type="text" value="192.168.7.255"/>

<i>Syslog Level:</i>	<input type="text" value="DEBUG"/>	
<i>NTP Server:</i>	<input type="text" value="time.nist.gov"/> (URI or IP address)	
<i>Distinctive Ring Tone:</i>	Custom ring tone 1, used if incoming caller ID is	<input type="text"/>
	Custom ring tone 2, used if incoming caller ID is	<input type="text"/>
	Custom ring tone 3, used if incoming caller ID is	<input type="text"/>
<i>Disable Call-Waiting:</i>	<input type="checkbox"/> No <input type="checkbox"/> Yes	
<i>Use Quick IP-call mode</i>	<input type="checkbox"/> No <input type="checkbox"/> Yes	
<i>Lock keypad update:</i>	<input type="checkbox"/> No <input type="checkbox"/> Yes	(configuration update via keypad is disabled if set to Yes)
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>		

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Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.

Voice Frames per TX	<p>This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the max. size of Ethernet packet is 1500 byte (or 120k bit) so user should be aware that there IS a limit there). When setting this value, the user should be aware of the requested packet time (ptime, used in SDP message) as a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time.</p> <p>e.g., if the first codec is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first codec chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the IP phone will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p> <p>Please be very careful when massage those parameters. By adjust this, user also get jitter buffer changed accordingly. BT-100 phone has patent dynamic jitter buffer handling algorithm. The jitter buffer range from 20 ~ 200 ms.</p> <p>Incorrect setting will affect voice quality so do not touch the parameter if not understand and most of the case the default value will work in GS products.</p> <p>Please refer to the Codec FAQ in our website for more technical details: http://www.grandstream.com/FAQ-Codec.pdf</p>
Video Frame Rate	Default value is 15 frames/second. User adjustable based on network condition.
Video bit rate (kbps)	Default value is 128 kbps. User adjustable based on network environment.
Video fit to Screen	When the received image size is bigger than the LCD resolution (QVGA), the GXV3000 can scale the image to fit the LCD or crop it.
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	This contains the value used for layer 2 VLAN tag. Default setting is blank.
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows users to configure the “#” key to be used as the “Send” (or “Dial”) key. If set to “Yes”, pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this “#” key will then be included as part of the dial string to be sent out.

Local RTP port	This parameter defines the local RTP-RTCP port pair the GXV-3000 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.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXV-3000s are behind the same NAT.
Keep-alive interval	This parameter specifies how often the GXV-3000 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server.
Firmware Upgrade and provisioning	This radio button will enable GXV-3000 to download firmware or configuration file through either HTTP or local TFTP server. This is mutual exclusive choice.
Via TFTP Server	<p>This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the GXV-3000 will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 attempts before 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 will be verified and then saved into the Flash memory.</p> <p>Note: It is strongly recommended that user upgrade firmware locally in LAN environment if using TFTP to upgrade. Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device.</p>
Via HTTP Server	<p>The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/1.0.0.14 Here “:6688” is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80.</p> <p>Note: If Auto Upgrade is set to No, GXV-3000 will only do HTTP download once at boot up.</p>
Automatic Upgrade	<p>Default is Yes. Choose Yes to enable automatic HTTP upgrade and provisioning. In “Check for upgrade every” field, enter the number of minutes to enable the phone to check the HTTP server for firmware upgrade or configuration changes in the defined period of time.</p> <p>When set to No, the phone will only do HTTP upgrade and configuration check once at boot up. Used by ITSP. End user should NOT touch these parameters.</p>
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833.

Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	<p>Select the ATA 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:</p> <ul style="list-style-type: none"> • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components:</p> <p>GS_LOG: [device MAC address][error code] error message</p> <p>Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>
NTP server	This parameter defines the URI or IP address of the NTP (Network Time Protocol) server which is used by GXV-3000 to display the current date/time.
Distinctive Ring Tone	Customer Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY sound this ring tone when the incoming call is from the Caller ID, device will use System Ring Tone 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.

Quick IP Call Mode	<p>This model has the ability to dial an IP address under the same LAN/VPN segment by simply pressing the last octet in the IP address, without SIP server required.</p> <p>In the Advanced Settings page there is an option "Use Quick IP-call mode", by default it is set to No. When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask.</p> <p>#XX or #X are also valid so leading 0 is not required (but OK).</p> <p>eg. 192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or # 192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or # 192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #</p> <p>192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3</p> <p>Note:- If you have a SIP Server configured, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.</p>
Lock keypad update	<p>If this parameter is set to "Yes", the configuration changes via keypad are disabled therefore the phone is some kind of locked.</p>

5.2.4 Individual Account Settings

Three independent SIP accounts each has its own configuration page. Their configurations are identical. The following is a screen shot of SIP Account 2 settings.

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
Account Active:	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
Account Name:	<input type="text" value="NetCentrex"/> (e.g., MyCompany)
SIP Server:	<input type="text" value="sip.netcentrex.com"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text" value="proxy.netcentrex.com"/> (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="20001014084505168"/> (the user part of an SIP address)
Authenticate ID:	<input type="text" value="20001014084505168"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="password"/> (purposely not displayed for security protection)
Name:	<input type="text" value="4084505168"/> (optional, e.g., John Doe)
Use DNS SRV:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
User ID is phone number:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
SIP Registration:	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
Unregister On Reboot:	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
Register Expiration:	<input type="text" value="2"/> (in minutes. default 1 hour, max 45 days)
local SIP port:	<input type="text" value="5062"/> (default 5062)
NAT Traversal (STUN):	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
SUBSCRIBE for MWI:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
Proxy-Require:	<input type="text"/>
Voice Mail UserID:	<input type="text" value="4084505000"/> (User ID/extension for 3rd party voice mail system)
Send DTMF:	<input checked="" type="checkbox"/> in-audio <input type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
Early Dial:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	<input type="text"/> (this prefix string is added to each dialed number)

Enable Call Features:	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)																
Session Expiration:	<input type="text" value="180"/> (in seconds. default 180 seconds)																
Min-SE:	<input type="text" value="90"/> (in seconds. default and minimum 90 seconds)																
Caller Request Timer:	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Request for timer when making outbound calls)																
Callee Request Timer:	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (When caller supports timer but did not request one)																
Force Timer:	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Use timer even when remote party does not support)																
UAC Specify Refresher:	<input type="checkbox"/> UAC <input checked="" type="checkbox"/> UAS <input type="checkbox"/> Omit (Recommended)																
UAS Specify Refresher:	<input checked="" type="checkbox"/> UAC <input type="checkbox"/> UAS (When UAC did not specify refresher tag)																
Force INVITE:	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Always refresh with INVITE instead of UPDATE)																
Enable 100rel:	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No																
Account Ring Tone:	<input type="checkbox"/> system ring tone <input checked="" type="checkbox"/> custom ring tone 1 <input type="checkbox"/> custom ring tone 2 <input type="checkbox"/> custom ring tone 3																
Send Anonymous:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (caller ID will be blocked if set to Yes)																
Auto Answer:	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes <input type="checkbox"/> Intercom/Paging																
Preferred Vocoder: (in listed order)	<table border="0"> <tr> <td>choice 1:</td> <td><input type="text" value="GSM"/></td> <td>choice 5:</td> <td><input type="text" value="GSM"/></td> </tr> <tr> <td>choice 2:</td> <td><input type="text" value="G.723.1"/></td> <td>choice 6:</td> <td><input type="text" value="PCMA"/></td> </tr> <tr> <td>choice 3:</td> <td><input type="text" value="G.729A/B"/></td> <td>choice 7:</td> <td><input type="text" value="PCMU"/></td> </tr> <tr> <td>choice 4:</td> <td><input type="text" value="PCMU"/></td> <td>choice 8:</td> <td><input type="text" value="PCMA"/></td> </tr> </table>	choice 1:	<input type="text" value="GSM"/>	choice 5:	<input type="text" value="GSM"/>	choice 2:	<input type="text" value="G.723.1"/>	choice 6:	<input type="text" value="PCMA"/>	choice 3:	<input type="text" value="G.729A/B"/>	choice 7:	<input type="text" value="PCMU"/>	choice 4:	<input type="text" value="PCMU"/>	choice 8:	<input type="text" value="PCMA"/>
choice 1:	<input type="text" value="GSM"/>	choice 5:	<input type="text" value="GSM"/>														
choice 2:	<input type="text" value="G.723.1"/>	choice 6:	<input type="text" value="PCMA"/>														
choice 3:	<input type="text" value="G.729A/B"/>	choice 7:	<input type="text" value="PCMU"/>														
choice 4:	<input type="text" value="PCMU"/>	choice 8:	<input type="text" value="PCMA"/>														
Preferred Video Coder: (in listed order)	<table border="0"> <tr> <td>choice 1:</td> <td><input type="text" value="H.264"/></td> <td>choice 2:</td> <td><input type="text" value="H.264"/></td> </tr> </table>	choice 1:	<input type="text" value="H.264"/>	choice 2:	<input type="text" value="H.264"/>												
choice 1:	<input type="text" value="H.264"/>	choice 2:	<input type="text" value="H.264"/>														
Jitter Delay:	<input type="text" value="Medium"/>																
Enable Video:	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes																
H.264 payload type:	<input type="text" value="99"/> (between 96 and 127, default is 99)																
Special Feature:	<input type="text" value="Standard"/>																
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>																	

Account Active	This field indicates whether the account is active or not. The default value for the primary account Account 1 is Yes. The default values for the other three accounts are No.
Account Name	A name to identify an account which will be displayed in LCD.
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 GXV-3000 for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can provide solution for it.
SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has 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.
Authenticate Password	SIP service subscriber's account password for GXV-3000 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name which will be used for Caller ID display.
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 GXV-3000 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 GXV-3000 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 will be cleared on reboot.
Register Expiration	This parameter allows user to specify the time frequency (in minutes) that GXV-3000 refreshes 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	This parameter defines the local SIP port the GXV-3000 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.

NAT Traversal	This parameter defines whether the GXV-3000 NAT traversal mechanism will be activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the GXV-3000 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXV-3000 will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXV-3000 will attempt to 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 GXV-3000 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.
Subscribe for MWI:	Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail User ID	When configured, user will be able to dial voice mail server by pressing “MSG” button. This ID is usually the VM portal access number.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
Early Dial	Default is No. Use only if proxy supports 484 response.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Enable Call Features	Default is No. If set to Yes, Call transfer, Call Forwarding & Do-Not-Disturb are supported locally.
Session Expiration	<p>Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated.</p> <p>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.</p>
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.
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN internetworking is to be supported. A user’s wish to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
Account Ring Tone	There are 4 different ring tone that are defined: <ul style="list-style-type: none"> • System Ring Tone: when selected, all calls will ring with system ring tone. • Customer Ring Tone 1 to 3: when selected, GXV-3000 will ONLY play this ring tone for all the incoming calls for this account.
Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Auto Answer	When set to “Yes”, GXV-3000 will automatically switch to speaker when there is an incoming call.
Preferred Vocoder	The GXV-3000 supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B. User can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 8”.
Preferred Video Coder	Currently GXV-3000 supports only H.264 codec.

Jitter Delay	Select desired Jitter Buffer Delay
Enable Video	When set to Yes, video is enabled on calls, otherwise disabled.
H.264 payload type	Enter a desired value (96-127) for dynamic RTP payload type for H.264 codec. Default is 99.
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

5.2.5 Saving the Configuration Changes

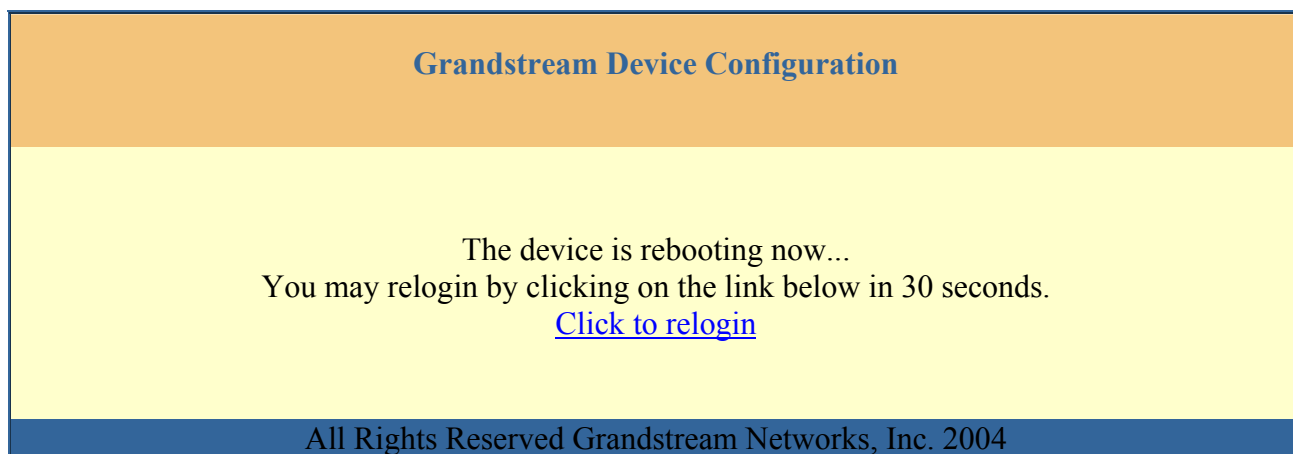
Once a change is made, the user should press the “Update” button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved:



User is recommended to power cycle the IP phone after seeing the above message.

5.2.6 Rebooting the Phone from Remote

The administrator of the phone can remotely reboot the phone by pressing the “Reboot” button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.



At this point, user can relogin to the phone after waiting for about 30 seconds.

5.3 Configuration through Central Provisioning Server

Grandstream GXV-3000 can be automatically configured from a central provisioning system.

When GXV-3000 boots up, it will send TFTP or HTTP request to download configuration file which is “cfg000b82xxxxxx”, where “000b82xxxxxx” is the MAC address of the GXV-3000.

The configuration file can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of GXV-3000 can easily manage the configuration and service provisioned for individual device from a central server remotely.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of GXV-3000. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual GXV-3000 for firmware upgrade, remote reboot, etc.

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting 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 server for further provisioning. Grandstream also provides GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration templates can be downloaded from http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please refer to the documentation of GAPS product or contact Grandstream Sales Department for more information.

6 Software Upgrade

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

6.1 Firmware Upgrade through TFTP/HTTP

To upgrade via TFTP or HTTP, user needs to select TFTP or HTTP upgrade method. “Upgrade Server” needs to be set to a valid URL of a 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/1.0.0.18

e.g. 168.75.215.189

There are two ways to set up the Upgrade Server to upgrade firmware, namely through the Key Pad Menu or via the GXV-3000’s Web configuration interface. To configure the Upgrade Server via Key Pad Menu options, select “System Config” from the Main Menu, then select “Upgrade Preferences”. Under this sub Menu, user can edit Upgrade Server in either an IP address format or FQDN format. Choose “Save and use TFTP” or “Save and use HTTP” to select upgrade method. And select “Reboot” from the Main Menu to reboot the phone.

To configure the Upgrade Server via the Web configuration interface, open up your web browser and enter the IP address of GXV-3000. Enter the admin password to get into the web configuration interface. In the ADVANCED SETTINGS page, enter the Upgrade Server’s IP address or FQDN in the “Upgrade Server” field. Select TFTP or HTTP upgrade method. Once these settings are entered, user needs to update the change by clicking the “Update” button. Then “Reboot” or power cycle the phone, the firmware files will be fetched upon booting up.

If the configured upgrade server is found and a new code image is available, the GXV-3000 phone will attempt to retrieve the new image files by downloading them into the phone’s SRAM. During this stage, the GXV-3000 phone’s LCD will show firmware file downloading process. Upon verification of checksum, the new code image will be saved into the Flash. If firmware upgrade fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXV-3000 phone will stop the upgrading process and simply boot using the existing code image in the flash.

Firmware upgrading may take as long as 45 minutes over Internet, or just 5 minutes on a controlled LAN. It is strongly recommended to conduct firmware upgrade in a controlled LAN environment whenever possible.

For users who do not have 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/her LAN to do firmware upgrading.

A free Windows version TFTP server can be downloaded from:

<http://support.solarwinds.net/updates/New-customerFree.cfm>.

Our latest official release can be downloaded from:

<http://www.grandstream.com/y-firmware.htm>

Unzip the file and put all of them under the root directory of the TFTP server. Put the PC running the TFTP server and the GXV-3000 phone in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the phone's web configuration page, configure the Firmware Server Path with the IP address of the PC, update the change and reboot the unit.

User can also choose to download the free HTTP server from <http://httpd.apache.org/> or just use Microsoft IIS web server

NOTE:

- When GXV-3000 phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the GXV-3000 phone. This file is for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

```
TFTP Error from [IP ADDRESS] requesting cfg000b82023dd4 : File does not exist  
Configuration File Download
```

NOTE:

- **After flashing the firmware, a power cycle is required. Please power cycle the phone after the downloading process finished and when the LCD becomes pale or white.**

6.2 Configuration File Download

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. “Upgrade Server” is the TFTP or HTTP server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format.

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 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 configuration template of the firmware.

When Grandstream Device boots up or reboots, it will issue request for configuration file named “cfgxxxxxxxxxxxx”, where “xxxxxxxxxxxx” is the MAC address of the device, i.e., “cfg000b820102ab”. The configuration file name should be in lower cases.

6.3 Managing Firmware and Configuration File Download

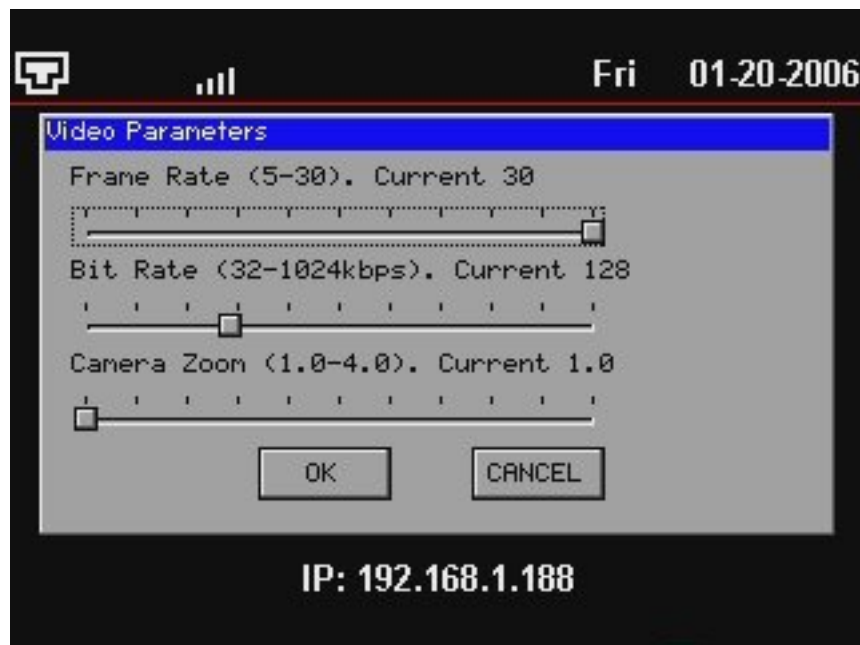
When “Automatic Upgrade” is set to “Yes”, Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check with either Upgrade Server. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

7 Auxiliary Ports

7.1 USB 2.0 Port

When a USB device is plugged into one of the two USB 2.0 ports, an icon will show up in the lower part of the LCD to illustrate the connected device. Following is the snap shot of the LCD when an USB device is plugged into the interface:

Figure 7-1:



(Remark: This screen shot is NOT correct, just put here for space occupation temporarily)

7.1.1 Capture pictures via USB port

When a USB flash drive is plugged into the USB port and icon showed up to illustrate the drive is ready, user can snap shot or capture the picture showed on the LCD and save it into the flash disk.

Here are the procedures: When the USB flash disk device is ready, whether the video phone is in camera local loopback mode or in a live video call, just press “OK” button once, the LCD will freeze 1 or 2 seconds, the captured picture will be saved in the disk using a format like: “**gxv_xxxx.yuv**”.

Only one picture can be saved at one time.

User needs special decoding software to see the captured pictures stored in the USB flash disk. A free software called XnView can be downloaded from:

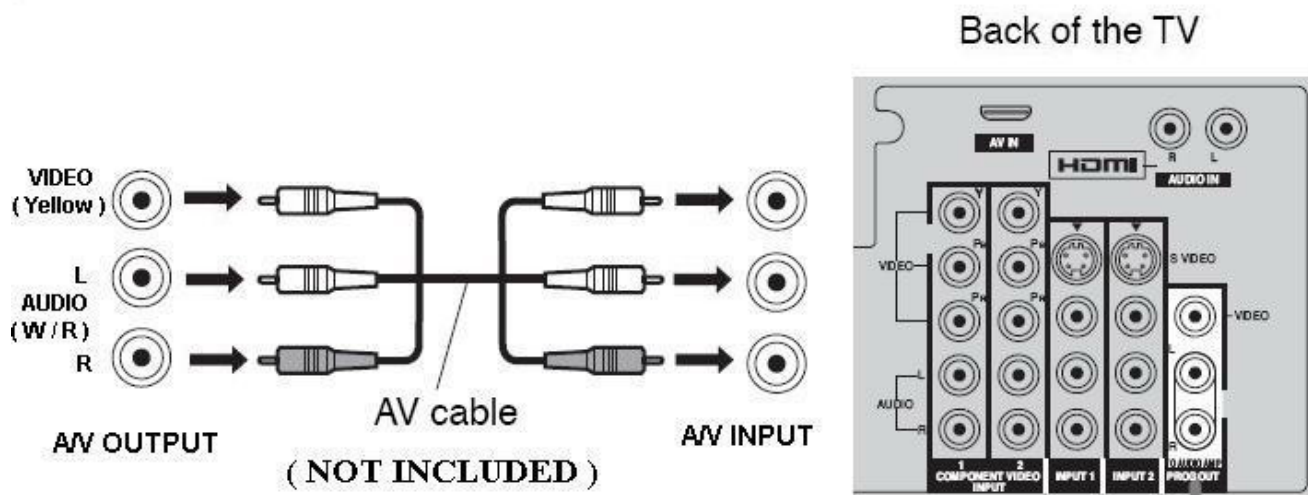
<http://perso.wanadoo.fr/pierre.g/xnview/enxnview.html>

Once this software installed, please choose file type “**YUV - YUV 4:2:2**” to open and see the captured pictures.

7.2 RCA style stereo audio & composite video output

The picture below shows the connection of GXV-3000 to TV set which allows use to watch the video and hear the audio on home TV set.

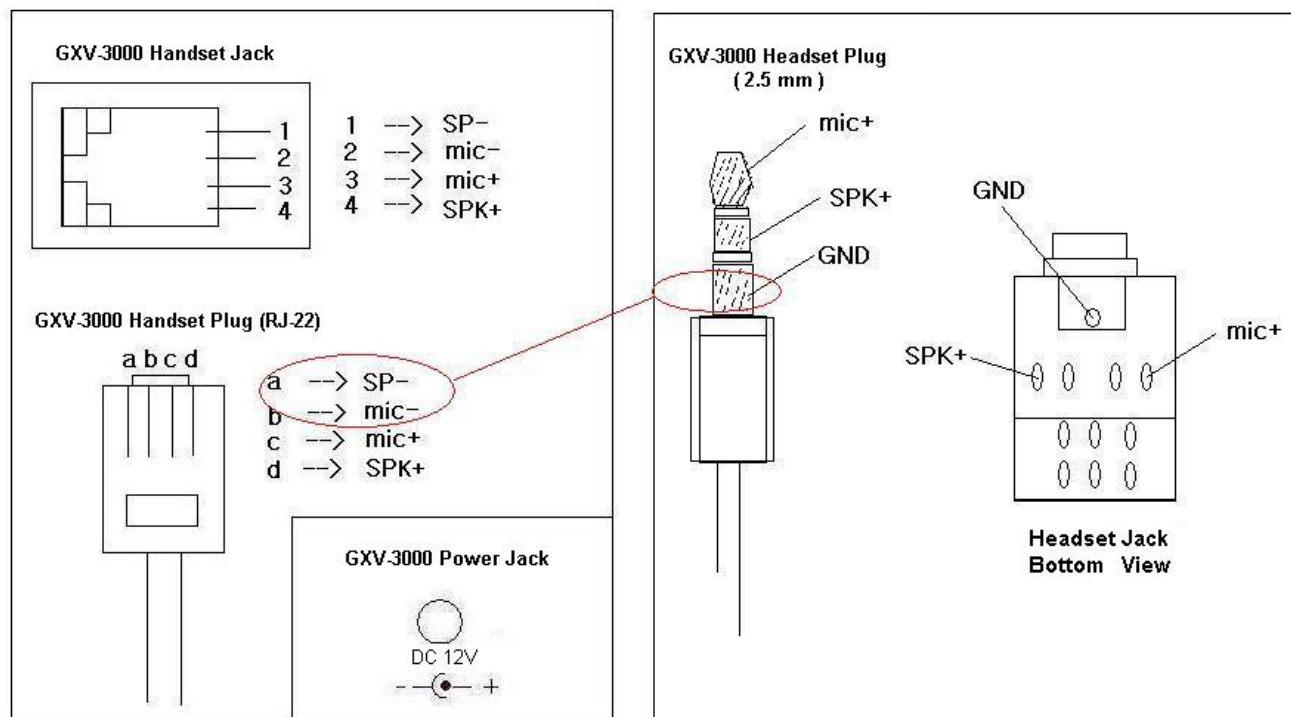
Figure 7-2:



7.3 Headset Jack

The picture below shows the handset and headset connectors' wiring schema of GXV-3000.

Figure 7-3:



As show in the schema, the left side is pin assignment of a RJ-22 handset plug; while the right side is showing a normal 2.5mm headset plug. If use want to connect the phone to external Speaker or Intercom/Paging system, necessary adaptors are required from local electronic stores.

8 Restore Factory Default Setting

WARNING !!!

*Restore 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.*

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:

Press “OK” key to bring up the key pad configuration UI menu, select “System Config”, press “OK” to enter submenu, select “Factory Reset” (Please refer to Table 5-1 of keypad flow chart)

Step 2:

Key in the MAC address which printed on the bottom of the sticker. Please use the following mapping:

0-9: 0-9

A: 22 (when pressed 2 twice, the “A” letter will show on the LCD)

B: 222

C: 2222

D: 33

E: 333

F: 3333

For example, if the MAC address is 000b8200e395, it should be key in as “0002228200333395”.

NOTE:

- If there is 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”.

Step 3:

Press the “OK” key again to move the cursor to “OK” button then press “OK” key again to confirm. If the MAC address you input is correct, the phone will reboot now. Otherwise it will exist to previous keypad menu interface.

9 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 subsampling.

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 multicarrier 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 multiframe (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

Short for *Domain Name System* (or *Service* or *Server*), an [Internet](#) service that translates [domain names](#) into IP addresses

DID

Direct Inward Dialing

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 Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

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 com is the top level domain.

FXO

Foreign eXchange Office

An FXO 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 FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

An FXS device will allow any FXO 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 FXO devices (by an FXO 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.

H.264

H.264, MPEG-4 Part 10, or AVC, for Advanced Video Coding, is a [digital video codec](#) standard which is noted for achieving very high [data compression](#). It was written by the [ITU-T Video Coding Experts Group \(VCEG\)](#) together with the [ISO/IEC Moving Picture Experts Group \(MPEG\)](#) as the product of a collective partnership effort known as the Joint Video Team (JVT). The ITU-T **H.264** standard and the ISO/IEC **MPEG-4 Part 10** standard (formally, ISO/IEC 14496-10) are technically identical. The final drafting work on the first version of the standard was completed in May of 2003.

The intent of H.264/AVC has been to create a standard that would be capable of providing good video quality at bit rates that are substantially lower (e.g., half or less) than what previous standards would need (e.g., relative to [MPEG-2](#), [H.263](#), or [MPEG-4 Part 2](#)).

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 party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. 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

i.e. 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 [Internet protocol suite](#). Using UDP, programs on networked computers can send short messages known as [datagrams](#) to one another. UDP does not provide the reliability and ordering guarantees that [TCP](#) 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 [speech processing](#) wherein, the presence or absence of human speech is detected from the audio samples.

VLAN

A virtual [LAN](#), known as a VLAN, is a logically-independent [network](#). Several VLANs can co-exist on a single physical [switch](#). It is usually refer to the [IEEE 802.1Q](#) tagging protocol.

VoIP

Voice over IP

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.