

Tone	disabled.
Disable Direct IP Call	The default setting is No. If set to Yes, Direct IP calling will be disabled.
Offhook Auto Dial	Configure a User ID/extension to dial automatically when the phone is taken offhook. By default, the phone will use the first account to dial out.

Advanced Settings/Video Settings

Enable Motion Detection	The phone will exit idle/screensaver mode if motion is detected. The default setting is Yes.
Video Frame rate	The Default value is 15 frames/second. The video frame rate is adjustable based on network conditions. Increasing the frame rate will increase the amount of transferred data significantly therefore consuming more bandwidth. Lack of bandwidth will impair the video due to packet loss.
Video Packet Size	The Default value is 1400, range from 100 to 1400. It is recommended to use 600~800 if you have an Outbound Proxy or Media Gateway.
Video Rate Control	Frame, TMN8 or GOP. The Default setting is Frame. TMN8 is good for bandwidths larger than 384kbps
Video Frame Skipping	Skips bad video frames as they are received The Default setting is No.
I-Frame Reference Only	When enabled, all P frames will only have reference to previous I frames. This method may improve video quality when the network has heavy packet loss. It will have modest increase in bandwidth usage. The Default setting is No.
Packetization-Mode	The packetization mode (0 or 1) for the H.264 video packets. The default setting is 0.
Redundant P-frame	This setting allows the phone to send redundant P-frames of I frames. When an I frame is delayed or has packet loss, the redundant P-frames will be used instead. It may make video phone call less prone to errors and thus make the video stream more robust towards error propagation, but may increase bandwidth usage. The default setting is set to No.
Adaptive MB Intra Refresh	This parameter specifies whether Adaptive MB Intra Refresh is used or not. It may protect most important MBs (macro blocks) subjected to packet loss and to avoid error propagation. The intra-refresh rate is adapted according to the video packet loss rate. It may increase bandwidth usage. The default setting is set to No.
Video Packet Loss Rate	Specifies the video packet loss rate for the Adaptive MB Intra Refresh. Users can choose from: Less than 5%, Less than 10%, Less than 15%, Less than 20%, Less than 25%. The default setting is set to Less than 10%.

Advanced Settings/Ring Tone

Call Progress Tones: Dial Tone Ring Back Tone	Using these settings, users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Frequencies should be configured with known values to avoid
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Busy Tone Reorder Tone Confirmation Tone Call-Waiting Tone PSTN disconnect Tone	uncomfortable high pitch sounds. Syntax: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing (“On time” in ‘ms’) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Default Ring Cadence	This defines the ring cadence for the phone. The default setting is: c=2000/4000;

MAINTENANCE PAGE DEFINITIONS

Maintenance/Network Settings

Address Type	This selects the type of IP address assigned: DHCP, PPPoE or Static IP.
PPPoE	When PPPoE is selected, the user needs to enter the following: PPPoE Account ID, PPPoE Password, PPPoE Service Name.
Static IP	When Static IP is selected, the user needs to enter IP address, Subnet Mask, Default Gateway, DNS Server 1 and DNS Server 2.
Alternate DNS Server	This field sets the alternate DNS server for the user.
Layer 3 QoS	This field defines the layer 3 QoS parameter. It is the value used for IP Precedence, Diff-Serv or MPLS. The Default value is 0.
Layer 2 QoS 802.1Q/VLAN Tag	This field contains the value used for layer 2 VLAN tagging. The Default value is 0.
Layer 2 QoS 802.1p Priority Value	This assigns the priority value of the Layer2 QoS packets. The Default value is 0.
802.1x Mode	This field sets 802.1x Mode. The default value is disabled. If EAD-MD5 is selected, users will be required to enter Identity and MD5 Password.
Proxy	This field sets HTTP Proxy, HTTPS Proxy, FTP Proxy or No Proxy. Using HTTP Proxy, other protocols can be applied by checking the box “Apply to Other Protocol”.

Maintenance/Time Settings

NTP Server	This parameter defines the URL or IP address of the NTP (Network Time Protocol) server. The GXV3174 phone may obtain the date and time from the server. The default setting is ntp.ipvideotalk.com.
DHCP Option 42 override NTP server	Select Yes for the user to allow DHCP Option 2 to override the NTP server if there is one on the LAN. The default setting is NO.
Time Zone	This parameter controls the date/time display according to the specified time zone.
Time Display Format	This parameter sets time display format to 12 Hour or 24 Hour

Date Display Format	This parameter sets date display format to YY-MM-DD or MM-DD-YY or DD-MM-YY.
Self-Defined Time Zone	<p>This parameter allows the users to define their own time zone.</p> <p>Syntax: std offset dst [offset], start [/time], end [/time] Default is set to: MTZ+6MDT+5,M3.2.0,M11.1.0, MTZ+6MDT+5 This indicates a time zone with 6 hours offset with 1 hour ahead which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east. M3.2.0,M11.1.0 The 1st number indicates Month: 1,2,3..., 12 (for Jan, Feb, ..., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday...) The 3rd number indicates weekday: 0,1,2,...,6(for Sun, Mon, Tues,...,Sat) Therefore, this example is the DST which starts from the second Sunday of March to the 1st Sunday of November.</p>

Maintenance/Web and Telnet Access

Disable Telnet	The default value is No. If set to YES, the GXV3174 will not allow any telnet access to the phone.
Access Method	This defines the access method for web access: HTTP or secure HTTPS protocol.
Port	This defines the port for web access. By default, HTTP uses port 80 and HTTPS uses port 443. This field is for customizable web ports.
Admin Password	This defines the administrator password for web interface access. Only the administrator can configure the “Advanced Settings” and “Account x” pages. The password is case sensitive and the maximum password length is 25 characters.
User Password	This defines the user password for web interface access.

Maintenance/Upgrade

Lock Keypad for Update	The default value is NO. If set to YES, the keypad will be disabled from making any updates.
XML Config File Password	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name for the HTTP server.
HTTP/HTTPS Password	The password for the HTTP server.
Upgrade Via	This field allows the user to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
Firmware Server Path	Defines the server path for the firmware server. It can be different from the Configuration server which is used for provisioning.
Config Server Path	This is the server path for provisioning; it can be different from the firmware server.

Firmware File Prefix	This field enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone.
Firmware File Postfix	This field enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the phone.
Config File Prefix	This field enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted prefix will be downloaded and flashed into the phone.
Config File Postfix	This field enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
DHCP Option 66 override Server	The Default setting is YES. If DHCP option 66 is enabled on the LAN side, the TFTP server can be redirected. Please be very careful when configuring this as the redirection could break the phone if this happens during the firmware upgrade.
Automatic Upgrade	The default value is NO. Choose "YES" to enable automatic HTTP upgrade and provisioning.
Period time of upgrade checking	Specifies the time period to check for firmware upgrade. The default setting is 10080 minutes (7 days)
Hour of the day (0-23)	Defines the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Day of the week (0-6)	Defines the day of the week to check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Automatic Upgrade Rule	Defines the rules for automatic upgrade: Always Check, when F/W suffix/prefix changes, Skip the Firmware Check.
Authenticate Conf File	Authenticate configuration file before acceptance. The Default setting is NO.
Factory Reset	Restore to factory default settings. Users can choose from: Full Reset, VoIP Data Reset, and User Data Reset. After the selection is made, press the "Reset" button to factory reset the phone. NOTE: Users can use the backup/restore functionality of the phone prior to the factory reset.

Maintenance/Syslog

Syslog Server	The URL/IP address for the syslog server.
Syslog Level	Select the level of logging for syslog. The default setting is NONE. There are 4 levels: INFO, DEBUG, ERROR and WARNING. Syslog messages are sent based on the following events: 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).
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Maintenance/Debug

Capture Trace	Press START to start capturing a trace, and press STOP to abort the capture process.
View Trace	Press LIST to view the list of captured traces. It will show the time the trace was captured. Click on the name of the file to download and view the trace.
Enable Application Debug	When set to YES, the trace captured will contain debug messages. The default setting is NO.

Maintenance/Language

Language	Selects the language setting for the phone. The current version supports English, Traditional Chinese and Simplified Chinese.
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Maintenance/Network Manager

ACS URL	Enter URL for TR-069 Auto Configuration Servers (ACS)
ACS Username	Enter ACS username
ACS Password	Enter ACS password
Periodic Inform Enable	Enable periodic inform. By default, it is disabled
Periodic Inform Interval(s)	Enter periodic inform interval(s). By default, it is set to 300s.
Connection Request Username	Enter connection request username
Connection Request Password	Enter connection request password

Maintenance/OpenVPN Settings

OpenVPN Enable	This enables/disables OpenVPN functionality. The default setting is No. To use OpenVPN functionality, it requires user to have access to an OpenVPN server.
OpenVPN Server Address	The URL/IP address for the OpenVPN server.
OpenVPN Server Port	The network port for the OpenVPN server. By default, it is set to 1194.
OpenVPN CA	OpenVPN CA file (ca.crt) required by the OpenVPN server for authentication purposes. Press "Upload" to upload the corresponding file to the phone.
OpenVPN Client Certificate	OpenVPN Client certificate file (*.crt) required by the OpenVPN server for authentication purposes. Press "Upload" to upload the corresponding file to the phone.
OpenVPN Client Key	The OpenVPN Client key (*.key) required by the OpenVPN server for

authentication purposes. Press “Upload” to upload the corresponding file to the phone.

NOTE: To use OpenVPN functionalities, users must enable OpenVPN and configure all of the settings related to OpenVPN, including server address, port, OpenVPN CA, certificate and key. Additionally, the user must also set the SIP account to use “VPN” for the “Nat Traversal” (under Account-> Network Settings). Reboot the phone after these settings are saved for the changes to take effect. Once the phone connects to the OpenVPN server and the authentication is correct, the phone will show the VPN IP address under the Status-> Network Status, but it will still display the LAN IP on the LCD of the phone.

Maintenance/Device Manager (LCD)

Screen Saver Timeout (m)	This sets the lapsed idle time needed for the screen saver to appear. The default setting is 5 minutes.
Screen Saver Interval (s)xd	The screen saver is a picture slide show. This defines the frequency to change the picture on the screen for the screen saver. The default setting is 10 seconds.
LCD Auto Power Off Time (m)	This defines the lapsed idle time needed for the LCD to power off automatically in order to save power. The default setting is 10 minutes.

Maintenance/Device Manager (Headset)

Headset TX Gain (dB)	Transmission gain. It’s a headset setting to control the voice intensity. The default setting is 0dB.
Headset RX Gain (dB)	Receive gain. It’s a headset setting to control the voice intensity. The default setting is 0dB.

Maintenance/Device Manager (Camera)

Zoom mode	Select the desired camera zoom mode (Tele, optical, digital). Can adjust local video zoom either in mirror mode or during the call (on the fly) by pressing the Left or Right Arrow Key.
Exposure Setting	Select the desired Camera Exposure mode based on the surroundings. The default setting is set to Auto. The default setting is 8.
Color Mode	Select the desired Camera Color Mode (Color or Monochromatic). The default setting is set to Color.
White Balance	Select the desired Camera White Balance to be automatic or Fixed. The default setting is set to Auto.
Lens Correction	Correct Camera Lens noise. The default setting is set to YES. It is recommended to set this parameter to YES to reduce camera lens noise and improve video quality.
Flicker Control	Reduces flicker of image shot with a video camera caused by luminance fluctuation of an illuminating light source. Users can select 50Hz/60Hz depending on the frequency of the commercial electrical power. The default setting is set to Auto.

Maintenance/Device Manager (TV Port)

TV Output	Select the desired TV Output type (PAL or NTSC). NTSC 4:3 by default.
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SIP ACCOUNT CONFIGURATION PAGE DEFINITIONS

The GXV3174 has three lines that can be configured to accommodate independent SIP accounts. Every SIP account has an individual configuration page.

Account/General Settings

Account Active	Select YES to enable and use the Line.
Account Name	The account name for the user that is displayed on the LCD screen.
SIP Server	The IP address of the SIP server or the given domain name of the VoIP service provider.
SIP User ID	User account information provided by the VoIP service provider; normally similar to a telephone number or an actual telephone number.
Authenticate ID	The authenticate ID for the SIP user. It can differ or be the same as the SIP user ID.
Authenticate password	The password that the GXV uses to authenticate with the ITSP (SIP) server. After it is saved, this will appear as blank for security reasons. The maximum length is 25 characters.
Voice Mail User ID	When this is configured, the user can dial to the voicemail server using the MESSAGE button. This ID is normally the feature code for Voice Mail.
Name	The Caller ID that will be displayed for the account.
Tel URI	The default setting is "Disable". If the Video phone has an assigned PSTN number, this field should be set to "Enable". If "User=Phone" is set, a "User=Phone" parameter will be attached to the "From header" in the SIP request to indicate the E.164 number.

Account/Network Settings

Outbound Proxy	IP address or Domain name of the Outbound Proxy, or Media Gateway, or Session Border Controller. Used by the GXV3174 for firewall or NAT penetration in different network environments. If a symmetric NAT is detected, STUN will not work and ONLY an Outbound Proxy will work.
DNS Mode	The default is set to A Record. If the user wishes to locate the server by DNS SRV, the user may select SRV or NATPTR/SRV.
NAT Traversal	This setting decides whether the NAT traversal mechanism is activated. If it is set to "STUN" and STUN server is configured, the GXV3174 will route according the STUN server. In this mode, the STUN client embedded in the phone will communicate with the appointed STUN server to examine which type of Firewall/NAT setting is employed. If the type of NAT detected is Full Cone, Restricted Cone or Port-Restricted cone, the phone will try to use public IP addresses and port in all the SIP and SDP messages. If the "NAT Traversal" is configured to be "Keep-alive", the phone will send an empty SDP packet (without payload data) to the SIP server once in 20 seconds to keep the NAT port open. If VPN is used, users should select "VPN" for NAT Traversal.

	Users could also set “Auto” or “UPnP” according to the network configuration. If an outbound proxy server is used, please configure this to be “NAT NO”.
Proxy Require	Notifies the SIP server that the unit is behind a NAT/Firewall.

Account/SIP Settings

SIP Registration	Selects whether or not the GXV3174 will send a SIP Register message to the proxy; The default setting is YES.
Unregister On Reboot	If set to YES, the SIP user registration messages will be cleared on reboot. The default setting is YES.
Register Expiration (m)	This defines the expiration time of the registration in minutes. The default value is 50 minutes; the maximum value is 65535 minutes (approximately 45 days) and the minimum value is 5 minutes.
Wait Time Retry Registration (s)	The wait time for registration retry if registration fails. The default is set to 20 Seconds.
Local SIP Port	The local SIP port for sending and receiving data. The default settings for Account 1/2/3 are 5060/5062/5064.
Subscribe for MWI	Default is set to No. When selected, a SUBSCRIBE for Message Waiting Indication will be sent out periodically.
Session Expiration (s)	Session Expiration is the time (in seconds) at which the session is considered timed out if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
MIN-SE (s)	The minimum session expiration (in seconds). The default value is 90 seconds.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or select UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use the caller or proxy server as the refresher, or select UAS to use the phone as the refresher.
Force INVITE	The Session Timer can be refreshed using the INVITE or UPDATE methods. Select “Yes” to use the INVITE method to refresh the session timer
Caller Request Timer	The default setting is Yes. If Yes is selected, the phone will use the session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If Yes is selected, the phone will use the session timer when it receives inbound calls with session timer request.
Force Timer	If Yes is selected, the timer will be activated even if the other party does not support session timers. If No is selected, the timer will be activated only if the other party supports the timer. To disable the timer, Caller Request timer, Callee request timer and Force Timer should all be set to No.
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is very important in order to support PSTN internetworking. To invoke a reliable provisional response, the 100rel tag is appended to the value of the required header of the initial signaling messages. The default setting is No.
SIP Transport	Selects the SIP transport method. There are three types: TCP, UDP and TLS.

Symmetric RTP	Selects whether or not symmetric RTP is supported.
Support SIP Instance ID	Selects whether or not SIP Instance ID is supported.
Validate Incoming Messages	This configuration selects whether or not the incoming messages should be validated.
SIP T1 Timeout	This configures the timeout period for SIP T1. The default setting is 0.5 Seconds.
SIP T2 Interval	This configures the time interval for SIP T2. The default setting is 4 seconds.
Remove OBP from route	The SIP Extension notifies the SIP server that it is behind a NAT/firewall.

Account/Codec Settings

Enable Video	--YES: Accepts all incoming video, this is the default setting; --NO: video is disabled. --Enable/Disable during dialing: Dialing time choice to enable/disable video
DTMF	Assigns the mechanism to send DTMF. There are three supported modes: in audio (with DTMF as audio signals), through RTP and through SIP INFO.
DTMF Payload Type	Sends DTMF using RFC2833. The default is 101.
Preferred Vocoder	This products supports up to 8 different Vocoders: including G.711(a/μ)(PCMU/PCMA),GSM,G726-32,G.723.1,G.729A/B, G.722 and L16-256. When this has been configured, the system will select the preferred vocoder from the list and the same order is sent with the SDP messages. The user can change the preferred vocoder priority order by toggling with the up and down buttons. It is recommended to use the default setting "PCMU"; if other vocoders are chosen, there could be an audio transmission error.
Preferred Video Codec	Selects the preferred video codec from the available list. It is recommended to use H.264.
H.264Payload type	Selects the dynamic H.264 codec payload type (ranges from 96-127). The default setting is 99.
H.263+ Payload type	Selects the H.263+ codec payload type (ranges from 96-127). The default setting is 103.
L16-256 Payload type	Selects the L16-256 codec payload type (ranges from 96-127). The default setting is 103.
H.263 Encoder Resolution	Selects the resolution (CIF/QCIF) used for H.263 codec. The default resolution is CIF.
Silence Suppression	This controls the G723/G729 silence suppression/VAD. If set to YES, when no audio is detected, the phone will send a small number of VAD packets (instead of audio packets). If Set to NO, this feature is disabled. The default setting is NO.
Voice Frame per TX	This configures the number of voice frames transmitted per packet. (It is recommended to set this based on the Ethernet IS packet limitation – 1500 Byte or 120kb/s). When configuring this, it should be noted that the "ptime" value for the SDP will change with different configurations here. This value is

	<p>related to the codec used and the actual frames transmitted during the in payload call.</p> <p>For example: if the codec is configured to be G.723 and “Voice Frames per TX” is set to 2. Then in the INVITE request, the “ptime” value for SDP will be 60 ms. This is because the G723 voice frame is 30 ms. Similarly, if the setting is 2 and the codec is G729 or G711 or G726, the “ptime” value for SDP in the INVITE request will be 20 ms.</p> <p>If the voice frames per TX configuration exceeds the maximum value allowed, the phone will choose the preferred codec and use the maximum value allowed.</p> <p>The maximum value is 64 frames.</p> <p>Adjusting these values will also change the jitter buffer. The phone has an algorithm for setting the jitter buffer and the jitter buffer rate is around 20-200ms.</p> <p>For end-users, it is recommended to use the default setting, as incorrect settings may influence the quality of audio.</p>
G723 Rate	This defines the encoding rate for G723 vocoder, and can be 6.3Kb/s or 5.3Kb/s. Please confirm with your service provider.
Jitter Buffer Type	Users can select either Fixed or Adaptive based on network conditions. The default is Adaptive.
Jitter Buffer Length	User can select Low, Medium, or High based on network conditions. The default is Medium.

Account/Call Settings

Dial Plan Prefix	Sets the prefix added to each dialed number.
Dial Plan	<p>Dial Plan Rules:</p> <ol style="list-style-type: none"> 1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , * , #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; <ol style="list-style-type: none"> a) xx+ - at least 2 digit numbers b) xx. - only 2 digit numbers c) ^ - exclude d) [3-5] - any digit of 3, 4, or 5 e) [147] - any digit of 1, 4, or 7 f) <2=011> - replace digit 2 with 011 when dialing g) - the OR operand <ul style="list-style-type: none"> • Example 1: {[369]11 1617xxxxxx} <p>Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617</p> <ul style="list-style-type: none"> • Example 2: {^1900x+ <=1617>xxxxxx} <p>Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers</p> <ul style="list-style-type: none"> • Example 3: {1xxx[2-9]xxxxxx <2=011>x+} <p>Allows any number with leading digit 1 followed by a 3 digit number, followed by any number between 2 and 9, followed by any 7 digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed.</p> <ol style="list-style-type: none"> 3. Default: Outgoing – {x+}

	<p>Allow any length of numbers.</p> <p>Example of a simple dial plan used in a Home/Office in the US: <code>{ ^1900x. <=1617>[2-9]xxxxxx 1[2-9]xx[2-9]xxxxxx 011[2-9]x. [3469]11 }</code></p> <p>Explanation of example rule (reading from left to right):</p> <ul style="list-style-type: none"> • ^1900x. - prevents dialing any number started with 1900 • <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically • 1[2-9]xx[2-9]xxxxxx - allows dialing to any US/Canada Number with 11 digits length • 011[2-9]x. - allows international calls starting with 011 • [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 <p>NOTE: In some cases where the user wishes to dial strings such as *123 to activate voice mail or other applications provided by their service provider, the * should be predefined inside the dial plan feature. An example dial plan will be: <code>{ *x+ }</code> which allows the user to dial * followed by any length of numbers.</p>
Early Dial	The default setting is NO. Use only if the proxy supports 484 response.
Refer to Use Target Contact	The default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target’s contact header information.
Auto Answer	The default setting is NO. If it is set to YES, the GXV3174 will play a “beep” sound and answer the phone automatically on the speaker.
Send Anonymous	If set to YES, the “From” header in the outgoing SIP messages will be set to anonymous to block the caller ID from displaying. The default setting is NO.
Anonymous Call Rejection	The default setting is NO; If set to YES, any anonymous call will be rejected (with 486 Busy message)
Special Feature	Different soft switch vendors have special requirements; therefore users may need to select special features to meet these requirements. Users can choose from Standard, CBCOM, RNK, China Mobile, ZTE IMS and Mobotix depending on their need. The default setting is Standard.
Enable Call Features	The default is set to YES. If it is set to NO, call features are disabled.
Call Forward Unconditional	Sets the number to forward to unconditionally.
Call Forward When Busy	Sets the number to forward to when busy.
Call Forward When No Answer	Sets the number to forward to when no answer.
Delayed Call Forward Wait Time (s)	The wait time before transferring the call to another extension or to voicemail; the default setting is 20 seconds.
No Key Entry Timeout (s)	Defines the time out duration on no key entry. Default setting is 4 seconds.
Ring Timeout (s)	Defines the time out duration on no answer. Default setting is 60 second.
Transfer on Conference	Defines whether or not the call is transferred to the other party if the

Hangup	initiator of the conference hangs up. Default setting is set to No.
Use # as Dial Key	Selects whether or not # is used as Send.
Account ring tone	This allows the user to configure the general ringtone used for this account. Users can choose from different ringtones from the dropdown menu. They can also upload mp3 files in Ringtone manager for their own ringtone.
Ring Tone	Users can select up to 3 distinctive ringtones to match incoming caller ID. Users can choose from different ringtones from the dropdown menu. They can also upload mp3 files in Ringtone manager for their own ringtone.

SAVING THE CONFIGURATION CHANGES

After users makes changes to the configuration, press the “Save” button in the Configuration Menu. The web browser will then display a message window to confirm the saved changes. Reboot the phone for the new settings to take effect if the settings that you have changed have a dot next to it.

REBOOTING FROM REMOTE LOCATIONS

Press the “Reboot” button at the top right-hand corner of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm with the user if they wish to reboot the phone or not. Click on “Yes” and the phone will reboot. Wait approximately 2 minutes to log in again.

LCD MENU AND FEATURES

The GXV3174’s user-friendly LCD MENU provides easy access to a host of features and convenient tools to help simplify the users’ business and personal life. Essentially, the GXV3174 can act like a personal assistant in many ways. In this section, we’ll explore the LCD MENU and Features and explain how users can take full advantage of the GXV3174’s capabilities.

CONTACTS

Users can easily manage all the contacts through CONTACTS. CONTACTS supports total records up to 500 entries and 20 groups with features such as add/delete/modify contacts, download phone book from the TFTP/HTTP server as well as the ability to save the phonebook to the PC. Select MENU-> CONTACTS and it will display in the figure below.

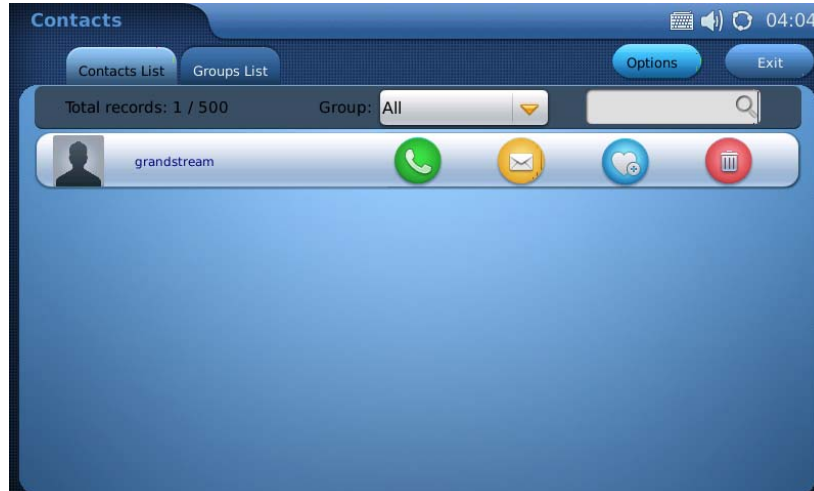







Figure 34: CONTACTS

Select “Contact List” or “Group List” tab to view/edit contact or group information.

In “Contact List”, users could select the group to view all the contacts in it. To search the contact by name, tap on the blank area besides  to input the name using the soft keyboard. In one entry in the contact list, press the icon  to make a call, press  to send short message, press  to add the contact to favorite and press  to delete. Select and Tap on one contact entry then the users could view the full contact information.

To add new contact, press the “Options” button and select option “New”. Then the screen will show as Figure 35. Every contact contains Work number, Home number, Mobile number and Fax number for a specific account. By tapping on the blank area in the field, the soft keyboard will be


activated for input. If the number is set to primary number, it will be dialed out when  is pressed in Figure 34. Thumbnail Image, ringtone could be customized to associate with each contact. Users could select from the dropdown list by pressing the “Down” arrow or browser to add ringtone file or picture from local or external storage. Press the “Down” arrow to select the group for the contact and press “Edit” to edit the group information. After finishing editing the contact, press “Save”.



Figure 35: Adding a new contact

To import/export contacts, in the screen shown as Figure 34, select “Options”->“Import/Export”. Then in the screen shown as Figure 36, select “Import” or “Export” in “Operation Mode”. There are three types of File to be selected: “XML”, “CSV” and “VCard”. For “Import”, users could select to “Clear The Old List” and “Replace Duplicate Items” in the imported files. If the operation mode is “Import”, users will need press on “Browse” button to specify the imported file from local or external storage device. Press “Import/Export” to finish the operation.



Figure 36: Import/Export contact list

In addition to manually adding each phonebook entry, the GXV3174 enables users to share and maintain the phonebook (by means of a XML phonebook) through the web easily. The XML phonebook must be stored on a HTTP/TFTP server. There are also applications where organizations will want to have a centralized directory server and have all phones in the organization synchronize with the server periodically.

To access this feature, in the screen shown as Figure 34, select “Options”->“Download”. Then in the screen shown as Figure 37, select “OFF”, “TFTP” or “HTTP” in “Download Mode”. Users could also select to “Clear The Old List” and “Replace Duplicate Items” in the downloaded phonebook. Tap on the blank field in “Download Server” to input the server URL using the soft keyboard. If “Download Interval (Min)” is set, the device will send request for the phonebook in the server for automatic update. After the setting, users could press “Download Now” for immediate download or press “OK” to finish the operation.



Figure 37: Download phonebook.xml

To delete all the contacts, users could select “Options”->“Delete All” in Figure 34.

CALL HISTORY

GXV3174 supports call record up to 100 dialed calls, 100 received calls and 100 missed calls. Users may access the phone’s call records through “MENU” -> “Call History”. See Figure 38.

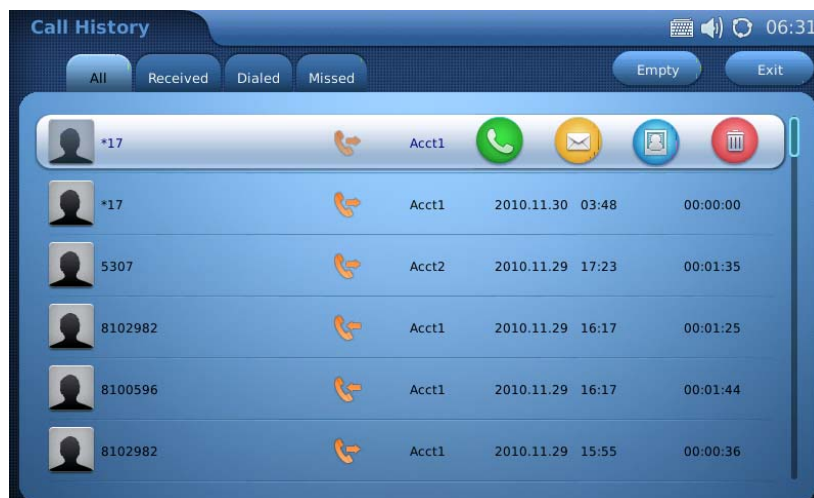






Figure 38: Call History

Users could select among the four tabs “All”, “Received”, “Dialed” and “Missed” to access the corresponding call record.

For each record, users can choose to Call , Message , Edit and Save to Contacts  or Delete  by pressing the icon. To delete all the call records in one page, tap on the “Empty” button.

MESSAGES

Messages is a text message feature in GXV3174. If two GXV3174s are registered on the same

SIP server, text messages can be sent/received between two parties. Similarly, if other IP products support this feature, the text message feature can also be used to send/receive messages between the device and the GXV3174. This application is accessible by selecting “MENU”-> “Messages”.

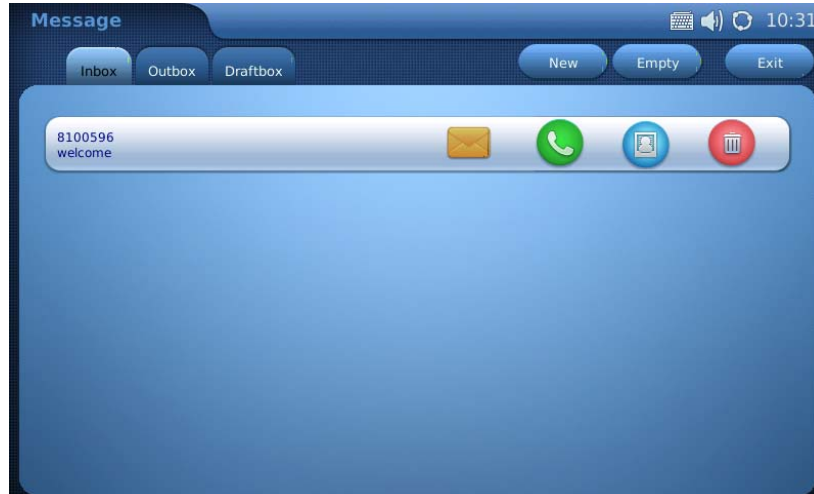






Figure 39: Message Inbox

In Messages, there are three categories “Inbox”, “Outbox” and “Draftbox” for users to select to view and manage the messages.

For each message, users press  to view the message, press  to dial the number out, press  to edit and save the contact, or press  to delete the message.

To send message, press “New” button. In Figure 40, edit the message by tapping on the blank area in the message content.

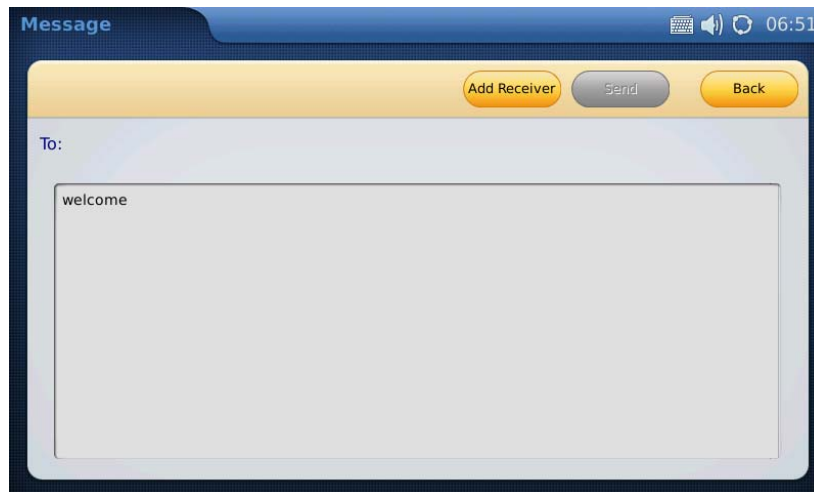


Figure 40: Edit new message

Then press “Add Receiver” to add the number in the window as shown in Figure 41. Users should choose the account from which the message will be sent out. After that, enter the phone number by tapping on the blank area in “Phone Number” field then press the activated “Add” button. Or

simply press “Find in Contacts” button to add the number from contact list.

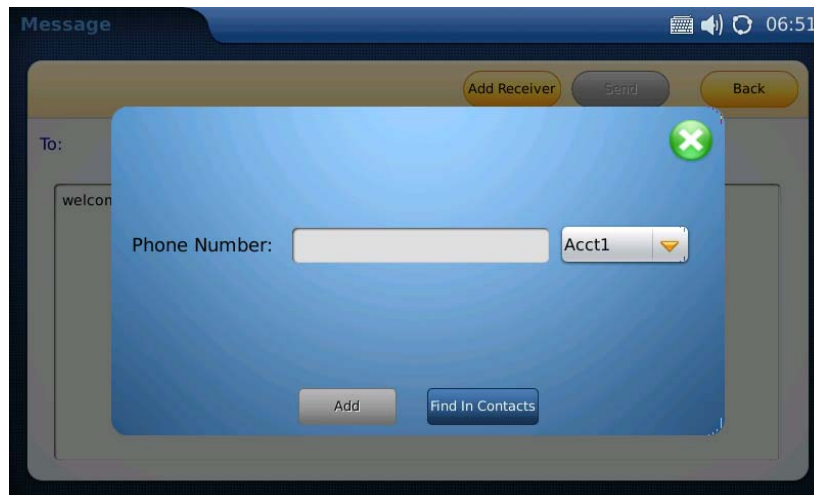





Figure 41: Add receiver's number

After pressing “Find in Contacts” button, users will see the screen as Figure 42 shows. The primary number for the contact will show in the list by default. If the users need to change to non-primary number, press  to add other numbers in the contact. Press  to add the number as message receiver. A yellow check will show on the left of the contact after pressing . Now tap on “Add” button and the screen will show in Figure 43.

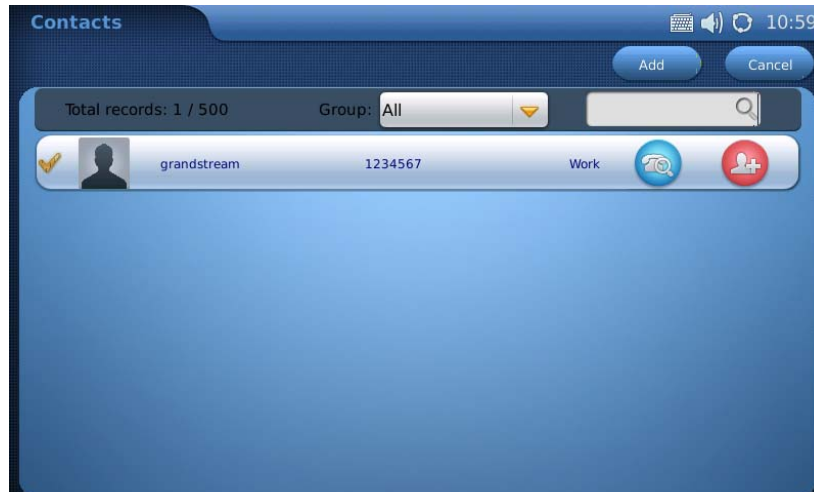


Figure 42: Add receiver's number from contact list

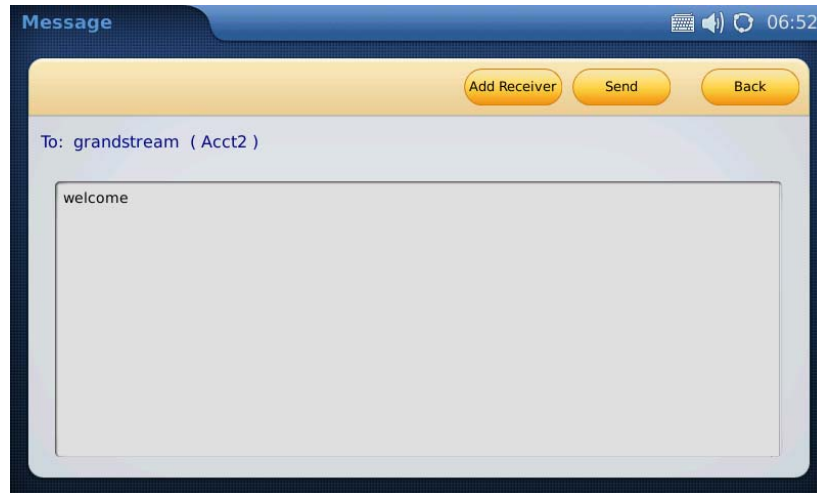


Figure 43: Send out message

The “Send” button will be activated after the message text and receiver are added successfully. Press “Send” to send the message out.

To view and reply message, select “Inbox” or press “View” in the incoming message alert window. (See Figure 44)

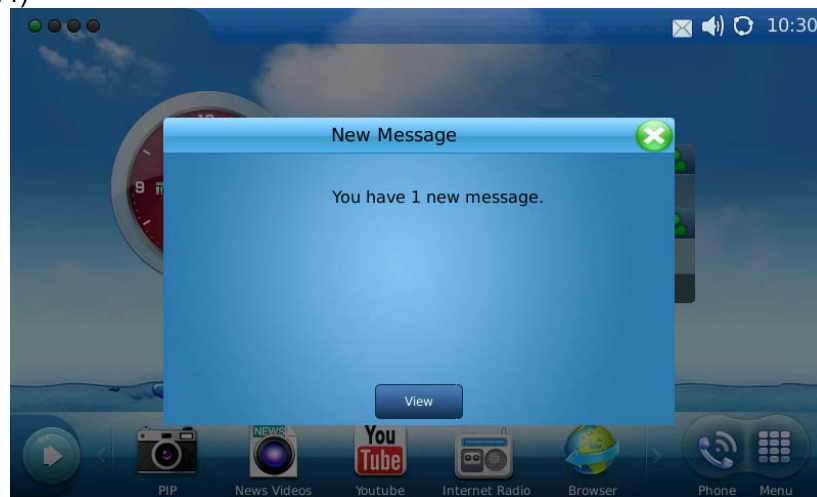


Figure 44: New message alert

To reply or forward the received message, press “Reply” or “Forward” in the Figure below.



Figure 41: Incoming message

To delete all the messages, press “Empty” button in Figure 39.

SOCIAL NETWORKS

- Google Voice

The GXV3174 IP Multimedia Phone supports Google Voice to enable the user to enjoy the social connections. Google Voice application in GXV3174 enables the users to make calls, send messages, manage contacts and account after logging in Google Voice account.

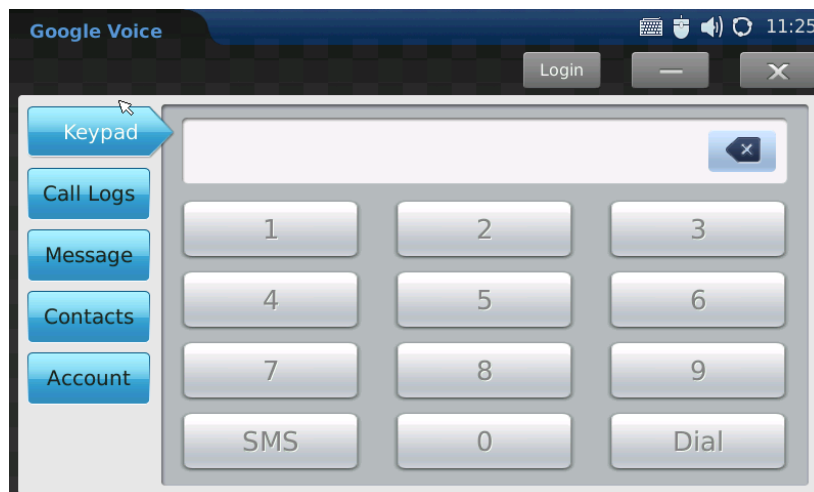


Figure 42: Google Voice in GXV3174

In Figure 42, press “Login” button and enter Email and Password via soft keyboard in Figure 43. Then Click “Login” in the login window.

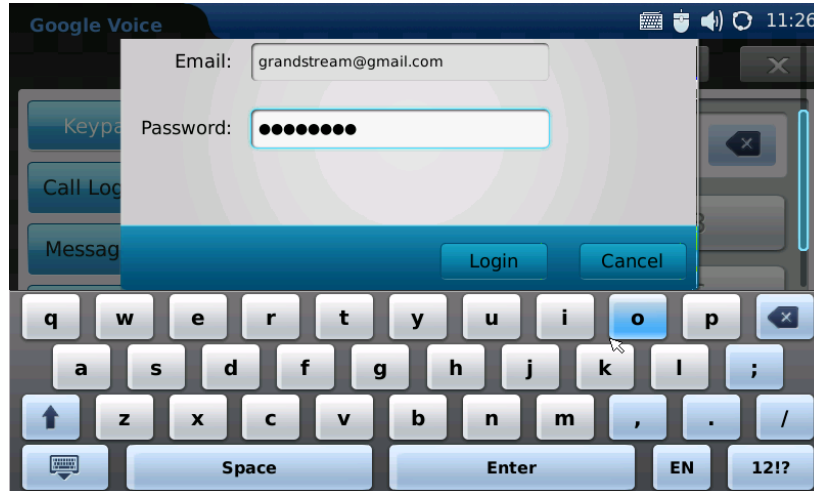



Figure 43: Login Google Voice Account



Figure 44: Using Google Voice in GXV3174

After logging in, users could tap on the number of the keypad shown in Figure 44. Then press “Dial” to make a call. Press “SMS” to send messages.

To view the call history, press “Call Logs” in the menu on the left of the screen and press the refresh icon . See Figure 45. Users will be able to dial out or send message to the contact in Call Logs by tapping on the selected entry in the displayed list.

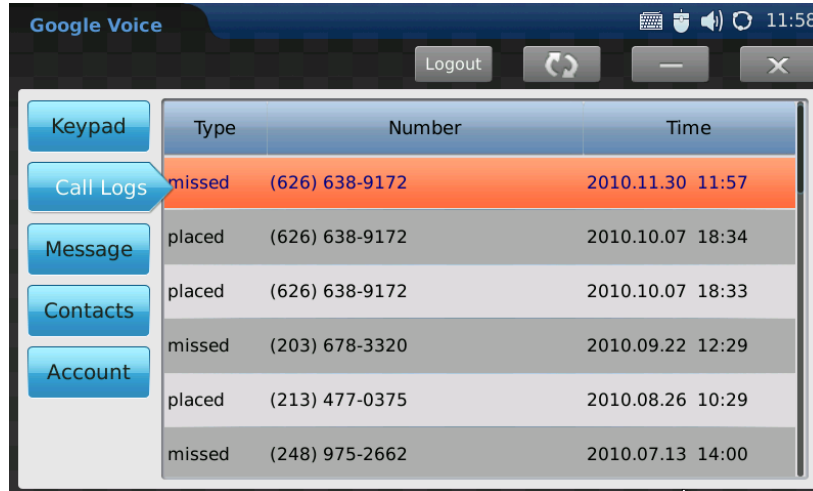



Figure 45: Google Voice-> Call Logs

To view messages, press “Message” and the refresh icon . See Figure 46. Select and press on one of the messages to reply the message or dial out. The maximum letters allowed in the message is 160.

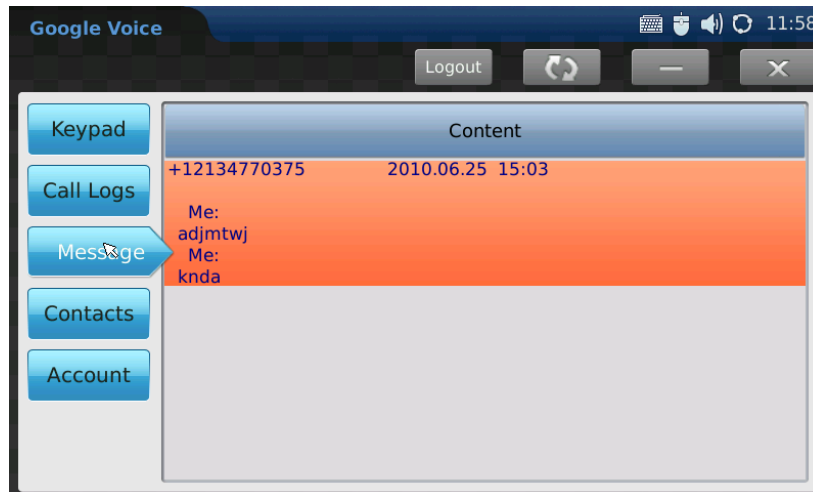


Figure 46: Google Voice-> Messages

To view the contacts set in google voice account, press “Contacts” and a list of contacts will show up in the screen. Users could also call or send messages to selected contact here by choosing one entry and tapping on it. See Figure 47.

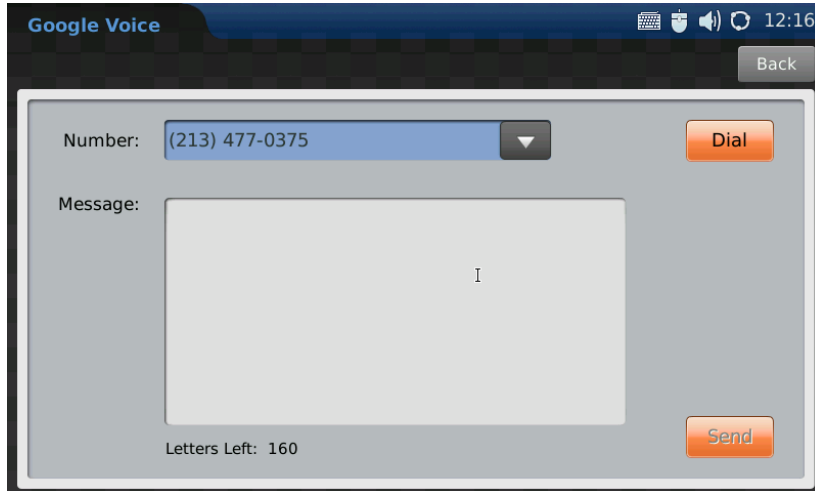


Figure 47: Google Voice-> Contacts

To setup Callback Number for the account, press “Account” and tap on the “Down” arrow to select the number from the dropdown list. See Figure 48.

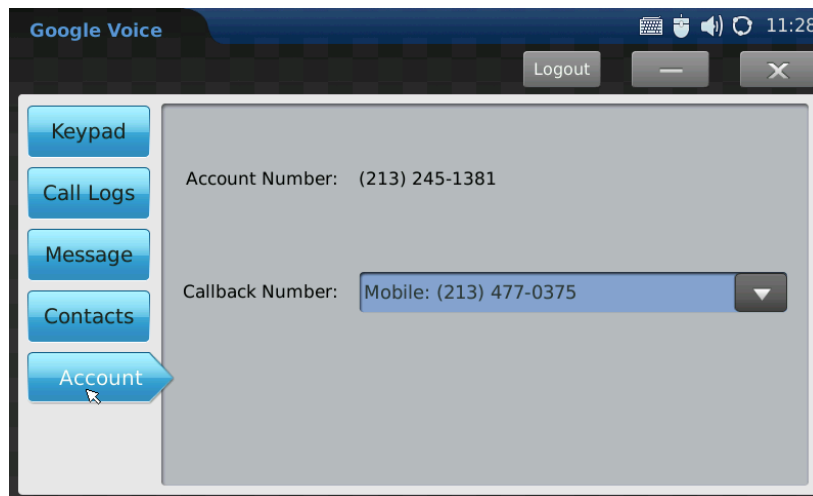




Figure 48: Google Voice Account Settings



To logout Google Voice account, press “Logout” and select “Yes” in the popping out message “Are you sure to logout?” Users could also minimize Google Voice and use other applications on GXV3174 at the same time without logging out by pressing icon . Press icon  to exit Google Voice.

BROWSER

The GXV3174 has a full HTML Web Browser for Internet browsing. The web browser is accessible by selecting “MENU”-> “Browser”. Tap on the blank area in navigation bar to enter the URL via soft keyboard and press “Enter” in the keyboard to visit the website. See Figure 49.



Figure 49: GXV3174 Web Browser

Users can view the web page by holding the finger and dragging in the touchscreen. Press icon  to move backward and  to move forward. Figure 50 shows the screen of the web browser while visiting Grandstream website.

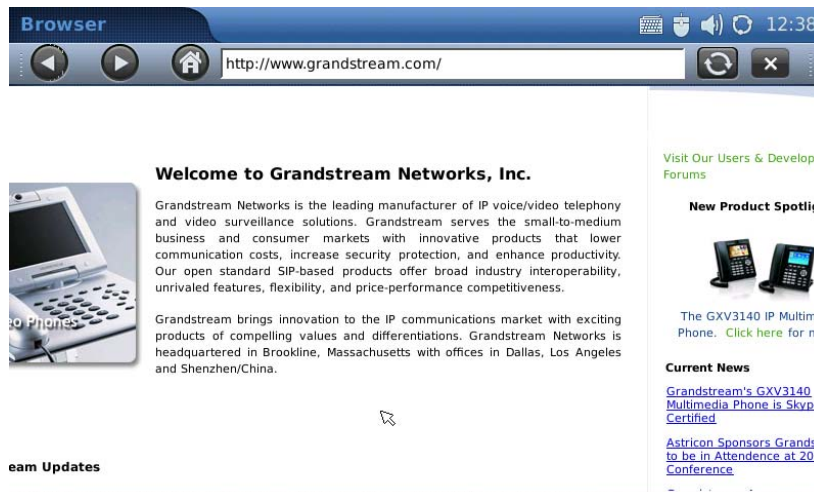




Figure 50: Using the GXV3174 Web Browser

While browsing the website, users can press icon  to refresh the webpage. To exit web browser, press .

MUSIC

To utilize the GXV3174's Online Music streaming application, users could access Internet Radio and Last.fm via "MENU"->"Music".

- **INTERNET RADIO**

The GXV3174 has a built-in Internet Radio application with thousands of radio channels from all over the world. The channels are divided into these categories: Favorite, Local, Talk, Music,

Sports and World.

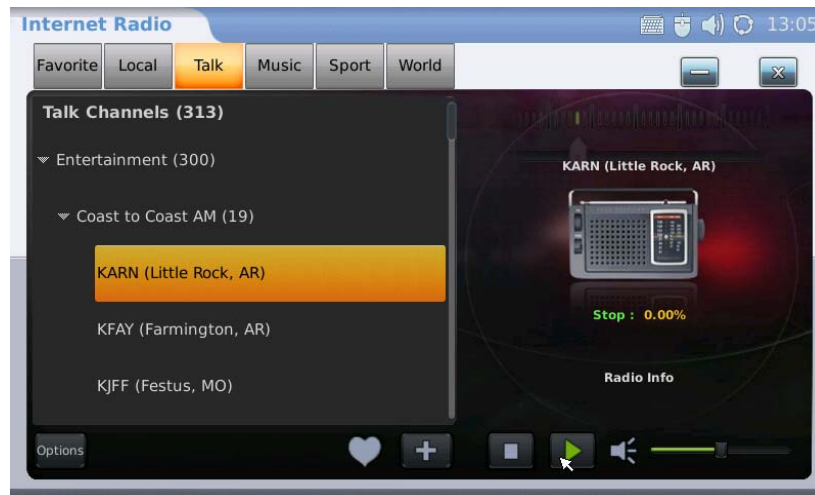






Figure 51: Menu->Music->Internet Radio

As shown in Figure 51, users can select one of the channels according to their preference. Then toggle through the channel list using scroll bar and select one channel. Press  or  to play or stop the selected program. Scroll the volume bar to adjust the volume. To change the audio output, press “Options” button down in the left corner. Then press “Output” and select “Speaker”, “Headset” or “Stereo”.

To add the channel to “Favorite” list, select one channel then press  icon to add it to “Favorite”. Users can also specify a channel to add in “Favorite” by pressing . In Figure 52, enter the Radio Name and Radio URL via soft keyboard by tapping on the blank field. Then press “OK”.

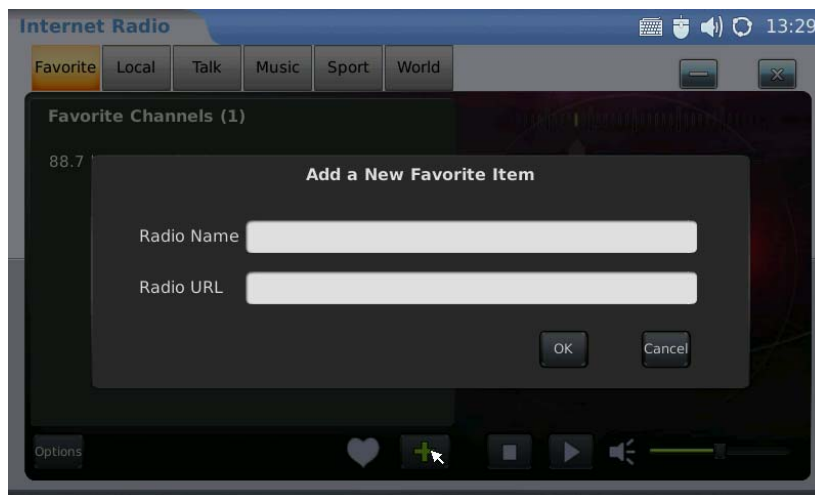


Figure 52: Internet Radio->Add Radio to Favorite

To close Internet Radio, press  to minimize the application and press  to exit.

- Last.fm

Last.fm is a social platform for users to select preferred music genre, share and listen to it online. Users must have a pre-existing account with Last.fm before listening to online music. Users could login the Last.fm account after registration via www.last.fm or on the phone following the instructions.

After the application loads, the username and password for Last.fm will be prompted. See Figure 53. If the users haven't registered a Last.fm account yet, press "Sign up for a Last.fm account" then the web browser will open and direct to <https://www.last.fm/join> for registration. Fill in Username and Password after registration then press "Login".

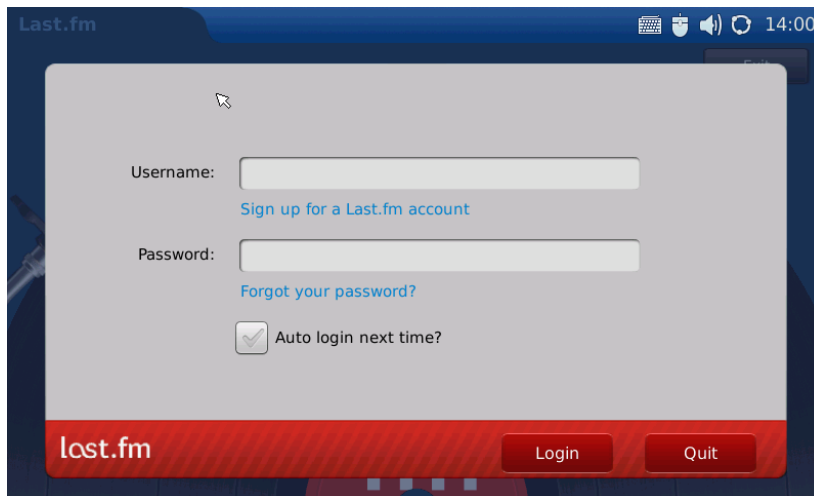


Figure 53: Last.fm->Login

If the users forget the password, press "Forgot your password". Then the web browser will open and direct users to last.fm website to retrieve the lost password. If "Auto login next time" is selected, the users will be automatically signed in after opening Last.fm application next time.



Figure 54: Last.fm->Search music

After logging in, users could search the music by "Artist" or "Tab" by entering the name in the








blank field above the playlist on the left. Press  to play the music in search result as shown in Figure 55. Users could select audio output among “Speaker”, “Headset” or “Stereo” by pressing “Output” button.



Figure 55: Last.fm->Play music

In Figure 55, users could press  to play the song and press  changing to the next song. Tap on the volume bar to adjust the volume. To add the song in “My Love Tracks”, press  then the song will be added the list under “My Love Tracks”. If  is selected, the song will not be played again. Users could also press  to add the song in playlist. To know more about the song being played, press  to access more related information. By pressing the small red arrow on the left, users could expand/hide the playlist. Press “Exit” button to logout Last.fm application.

VIDEO

- YOUTUBE

Online video streaming is available on the GXV3174 via integration with YouTube. Access Youtube application through “MENU”-> “Multimedia” -> “Youtube”.

All YouTube videos are viewable. There are some default video feeds for users to browse, which includes Standard Feeds (Top Rated, Most Popular and etc), Category (Music, Sports, Entertainment, News and etc), Shows, Movies and etc. (See figure 56) Press the thumbnail and select “Watch now” to watch the video.

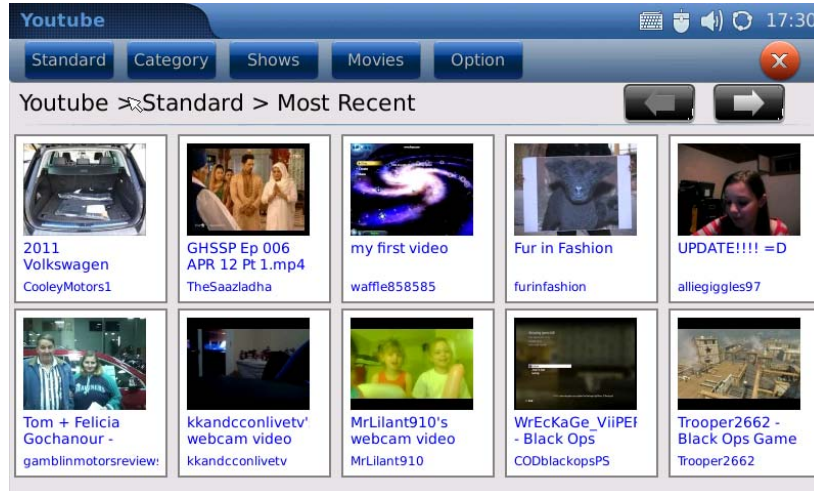


Figure 56: Youtube Application

Users can also upload their own videos to YouTube via GXV3174. To upload a video to YouTube, the user has to login with an existing YouTube account or register for a YouTube account first via Options->Login. See Figure 57.

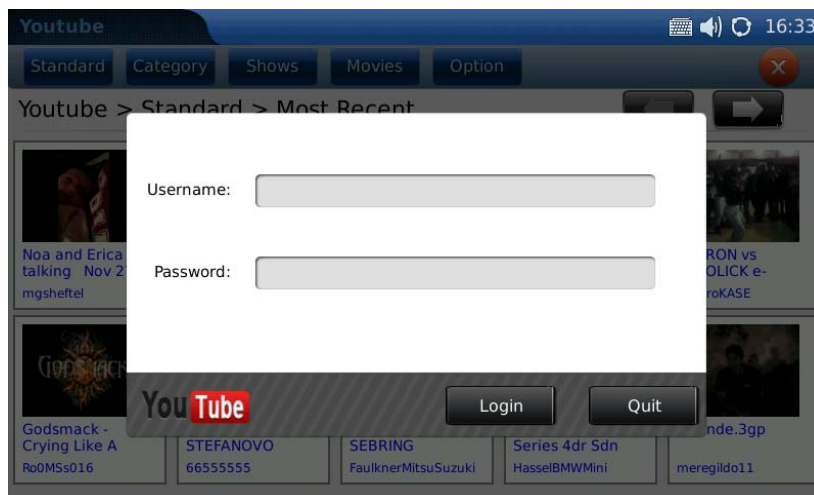


Figure 56: Youtube Account Login

After login is successful, select "Option"-> "Upload". In the screen as Figure 57 shows, fill in the required information (Title, Category, Keyword, Description), select the video file from local or external storage device and press "Submit".



Figure 57: Upload Videos to Youtube Account

Besides viewing videos from the built-in video feed, users can also search for videos according to keyword and preferences. This is done by pressing “Option”->“Search”. Then enter the keyword in “Search Word” and press “Search” button. (See figure 58)

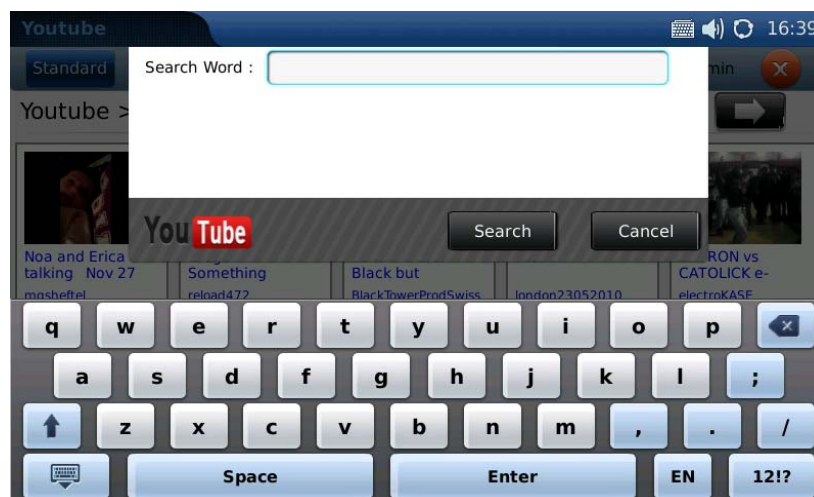


Figure 58: Search Online Videos

To add the video to “Favorite”, users need to login Youtube account. Then after select a video, press “Favorite” button (See Figure 59). Now users will be able to access to all videos marked as “Favorite” in “Option”->Favorite.

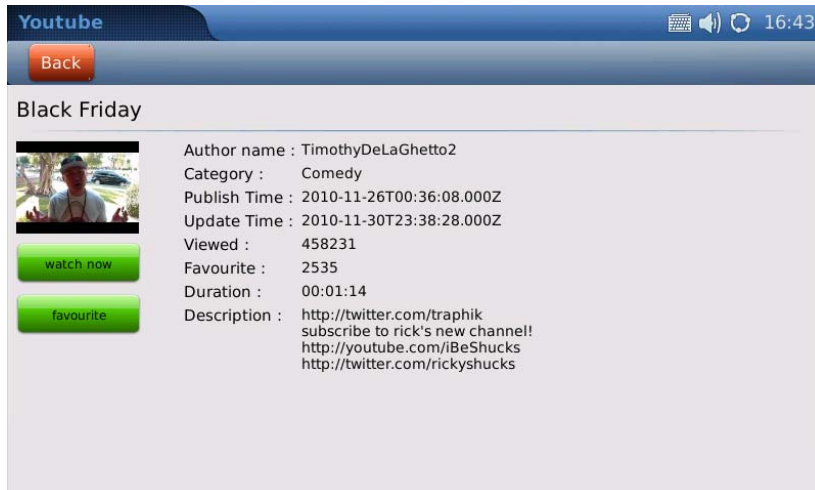


Figure 59: Add video to Favorite

SOFTWARE UPGRADE

The GXV3174 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. To configure a valid URL for TFTP or HTTP, the server name can be the FQDN or IP address format. Examples of valid URLs:

1. fw.ipvideotalk.com/gs
2. 72.172.83.110

There are two ways to setup a software upgrade server: Via Menu or the Web Configuration Interface.

UPGRADE VIA MENU

In "Menu"->"System Settings"->"Maintenance", users could configure the software upgrade server through the menu in the phone. In the "Upgrade" subpage, users may then select the upgrade mode and enter the IP address or FQDN of the Firmware server and the System Config server. After making the changes, press save to apply the new settings and reboot the device for upgrading.

UPGRADE VIA WEB BROWSER CONFIGURATION PAGE

Open a web browser and enter the IP address for the GXV3174. Next, log in with the administrator username and password. Under Maintenance, select "Upgrade and provisioning", enter the IP address or the FQDN (URL) of the upgrade server and choose to upgrade via TFTP or HTTP/HTTPS. Save the changes and reboot the unit.

At this moment, the LCD will display the progress of the software download/upgrade. Please do not disrupt or power down the unit.

NOTE: The software upgrade will last up to 5 minutes (through the LAN side) and may take longer when upgrading through the Internet (limited by the network bandwidth). Grandstream recommends downloading and upgrading the software in a controlled LAN environment.

NO LOCAL TFTP SERVERS

For users without a local TFTP server, Grandstream offers a NAT-friendly TFTP server. This enables users to download the latest software upgrades for their phone via this server. Please refer to the webpage: <http://www.grandstream.com/firmware.html>.

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free windows version TFTP server is available for download from <http://support.solarwinds.net/updates/New-customerFree.cfm> or <http://tftpd32.jounin.net/>.

Instructions for local firmware upgrade:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server.
2. Connect the PC running the TFTP server and the GXV3174 device to the same LAN segment.
3. Launch the TFTP server and go to the File menu -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server and configure the TFTP server in the phone's web configuration interface.
5. Configure the Firmware Server Path to the IP address of the PC.

6. Update the changes and reboot the GXV3174.

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

RESTORE FACTORY DEFAULT SETTINGS

WARNING: Restoring the Factory Default Settings will **delete** all configuration information on the phone.

Please **backup** or **print** all the settings before you restore to the factory default settings. Grandstream is not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

There are two methods to restore the GXV3174 to the factory default settings.

RESTORE TO FACTORY DEFAULT SETTINGS VIA LCD MENU

The steps to restore the phone to factory default settings through the LCD menu are as follows:

Step 1: Select MENU to access the phone menu and select System Settings.

Step 2: Select Maintenance and navigate to Upgrade page.

Step 3: Select the Factory Reset option.

Step 4: Press "OK". The phone will reboot and perform a factory reset.

After factory reset, users will need to do calibration before the phone's main screen show up.

RESTORE TO FACTORY DEFAULT SETTINGS VIA THE WEB INTERFACE

Step 1: Login to the system configuration page through a web browser and select "Maintenance". From the menu on the left, select "Upgrade".

Step 2: At the bottom of the "Upgrade" page, click on the "Reset" button for Factory reset after selecting the degree of factory reset (Full Reset/User Data Reset/VoIP Data Reset).

Step 3: Click on OK to restore the phone to factory settings.



EXPERIENCING THE GXV3174 MULTIMEDIA PHONE

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

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

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

Thank you again for purchasing Grandstream IP Multimedia phone, it is sure to bring convenience and color to both your business and personal life.

FCC Certification Mark

Warning

This device complies with Part 15 FCC specifications, and operation complies with the following conditions : (1) The device may not cause harmful interference, and (2) The device must tolerate any interference, including interference could lead to misuse;

Any change on equipment by users without the permission will result in the losing the right to use the equipment;

For a Class B digital device or peripheral, the instructions furnished the user shall include the following or similar statement, placed in a prominent location in the text of the manual:

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

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

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