GRANDSTREAM
CONNECTING THE WORLD

Grandstream Networks, Inc.

HT701/HT702/HT704

FXS Port

Analog Telephone Adaptors

User Manual

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CAUTION

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

WARNING

Please do not use a different power adaptor with your devices as it may cause damage to the products and void the manufacturer warranty.

GNU GPL INFORMATION

HT70X firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

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http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download

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CHANGE LOG

This section documents significant changes from previous versions of user guide for HT70X. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.9.4

- No major changes.

Firmware Version 1.0.8.2

- Upgrade the option Allow DHCP Option 66 to override server to [Allow DHCP Option 66 or 160]
- Added the options to enable/disable [3CX Auto Provision]
- Added the options to enable/disable [Disable Attended Transfer]
- Added the options to configure [SIP Registration Failure Retry Wait Time upon 403 Forbidden]
- Added the options to [Enable SIP OPTIONS Keep Alive] to use SIP OPTIONS instead of STUN as Keep-Alive messages.
- Added the options to enable/disable [Do Not Escape '#' as %23 in SIP URI]
- Added the options to configure [SIP OPTIONS Keep Alive Max Lost]
- Added the options to configure [Hold Target Before Refer] allows user to hold or not hold the phone call before refer.
- Added the options to enable/disable [Authenticate server certificate domain].
- Added the options to enable/disable [Authenticate server certificate chain]
- Added the options to configure [Trusted CA certificates]
- Added the options to enable/disable [Flash Digit Control]
- Added the options to enable/disable [Disable Connected Line ID]
- Added the options to configure [Conference URI]
- Added the options to configure [Fax Tone Detection Mode]
- Added the options to enable/disable [Crypto Life Time]
- Added the options to enable/disable [Enable Pulse Dialing]

Firmware Version 1.0.7.3

- Added the options to enable/disable NEC [Disable Network Echo Suppressor]

- Added the options to restrict outgoing call duration [Outgoing Call Duration Limit]
- Added the options to configure [Ring Frequency]
- Added the options to [Enable High Ring Power], to support device consumer more power from FXS port.
- Added the options to enable/disable [Generate Continuous RFC2833 Events].
- Added the options to configure [SIP REGISTER Contact Header Uses] it's WAN or LAN address.
- Added the options to [Disable Multiple m line in SDP].
- Added the options to configure [SIP Timer D].
- Added the options to configure [Country Specific Deployment]

Firmware Version 1.0.6.1

- Added the options to enable/disable [Always send HTTP Basic Authentication Information]
- Rearrange Layer 2 QoS setting options [Layer 2 QoS], and move [Layer 3 QoS] to FXS Port page

Firmware Version 1.0.5.2

- Added the option to define [Voice Frames per TX]
- Added the options to enable/disable [Use P-Preferred-Identity Header] and [Use Privacy Header]
- Added the option to enable/disable [Add Auth Header On Initial REGISTER]

Firmware Version 1.0.4.14

- Added the option to enable/disable [Allow DHCP Option 120(override SIP server)]
- Added the option [DTMF Caller ID Tone] to define DTMF call ID Start/Stop Tone

Firmware Version 1.0.4.8

- Added the option to enable/disable HTTP Access [HTTP Access]
- Added the option to enable/disable Authenticate incoming INVITE [Authenticate incoming INVITE]
- Added ability to configure the time of re-register before registration expired [Reregister before Expiration]
- Updated Table3 Advanced Definitions of the HT70X LEDs Pattern
- Added the option to enable/disable Use DNS to detect network connectivity [Use DNS to detect network connectivity]

Firmware Version 1.0.3.1

- Added the option to enable/disable hook flash [Enable Hook Flash]
- Removed firmware key from Advanced Setting page

GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/HT70X_web_gui.zip

1. Screenshot of Login Page
2. Screenshots of Status Page
3. Screenshots of Basic Settings Page
4. Screenshots of Advanced Settings Page
5. Screenshots of Profile 1 Page
6. Screenshots of Profile 2 Page
7. Screenshots of FXS Ports Page

WELCOME

The HT70X analog telephone adaptors provide transparent connectivity for analog phones and faxes to the world of internet voice. Connecting to any analog phone, fax or PBX, the HT70X are an effective and flexible solution for accessing internet-based telephone services and corporate intranet systems across established LAN and internet connections. The Grandstream handy tones HT70X are new additions to the popular handy tone ATA product family.

This manual will help you learn how to operate and manage your HT70X analog telephone adaptor and make the best use of its many upgraded features including simple and quick installation, 3-way conferencing, direct IP-IP Calling, and new provisioning support among other features. The HT70X are very easy to manage and configure, and are specifically designed to be an easy to use and affordable VoIP solution for both the residential user and the teleworker

CONNECT YOUR HT70X

Equipment Packaging

The HT70X ATA package contains:

- One HT70X Main Case
- One Universal Power Adaptor
- One Ethernet Cable
- One Vertical Stand (Only on HT702 and HT704 Packages)

Connecting the HT70X

The HT70X is designed for easy configuration and easy installation.   Configure the HT70X following the directions in the Configuration section of this manual.

1. Insert a standard RJ11 telephone cable into the Phone port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Insert the Ethernet cable into the Internet or LAN port of the HT70X and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
3. Insert the power adapter into the HT70X and connect it to a wall outlet.

The HT70X Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.   The HT70X VoIP features and functions are available using a regular analog telephone.

HT701

Display LEDs

Internet Port

(RJ-45 connector

10/100 Mbps)

Reset

Power

Supply

Phone

Display LED's (green)

Display LED's (green)

LAN Port

(RJ-45 connector

10/100 Mbps

with LEDS)

Reset

Power

Supply

Phone (RJ-11

HT702

Phone (RJ-11
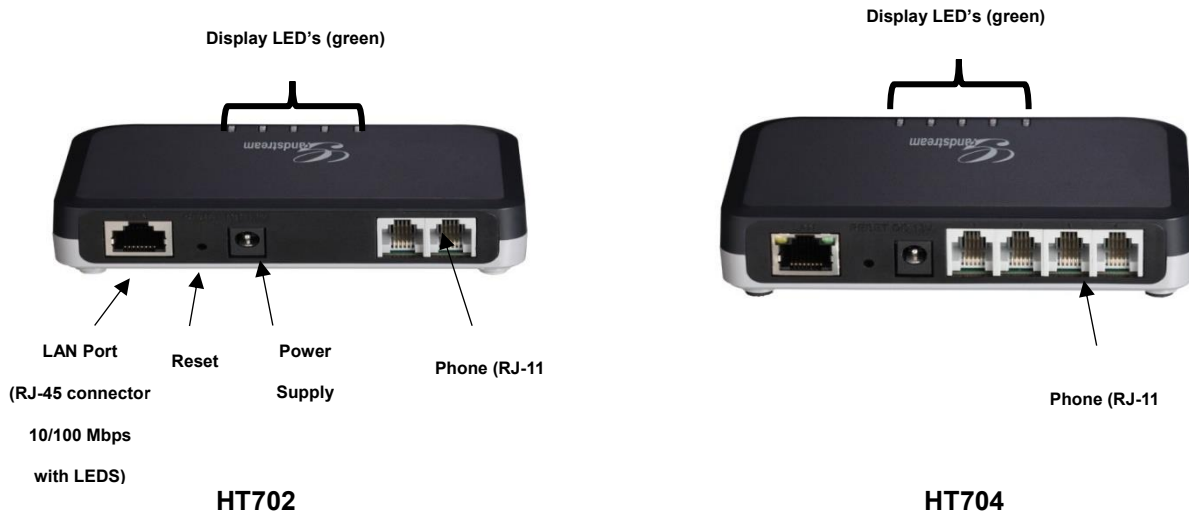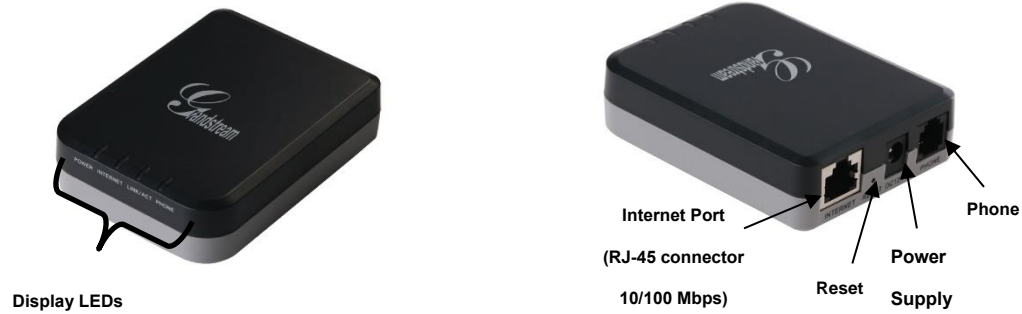
HT704

Figure 1: Diagram of HT70X

Table 1: Definitions of the HT70X Connectors

DC 12V	Power adapter connection
Internet Port (RJ-45)	Connect to the internal LAN network. **(HT701 Only)**
LAN Port (RJ-45)	Connect to the internal LAN network. **(HT702 and HT704 Only)**
RESET	Factory Reset button: Press for 7 seconds to reset factory default settings.
Phone Port(s) (RJ-11)	FXS port: to be connected to analog phones / fax machines.

There are four (4) LED buttons that help you manage the status of your Handy Tone **701**, and there are five (5) LED buttons that help you manage the status of your Handy Tone **702** and **704.**

Table 2: Basic Definitions of the HT70X LEDs Pattern

POWER LED	Indicates Power. Remains ON when power is connected
	HT702 Only)
Link/Activity LED	Indicates if There is Activity on the Internet Port **(HT701 and HT702 Only)**
PHONE LED	Indicate status of the respective FXS Ports-PHONE on the back panel: Unregistered – OFF Registered and Available – ON (Solid Green) Off-Hook / Busy – Blinking every second Slow blinking FXS LEDs indicates voicemail

NOTE: All LEDs display green when ON

Table 3: Advanced Definitions of the HT70X LEDs Pattern

LED-01	Device has normal power	Power	ON
LED-02	Power Error: Power is removed from the device or power supply with improper voltage is plugged in	Power	OFF
LED-03	Line X is registered normally to the sip providers network and is ready to make a call	Phone	ON
LED-04	Voice mail waiting for Line X	Phone	1sec ON / 3sec OFF
LED-05	Device has normal WAN connection and has obtained IP address	Internet	ON
LED-06	Internet link error: Device is powered up and ready to connect to the Internet but the WAN/INTERNET port is down	Internet	OFF
LED-07	Internet DHCP Error: Device is properly connected but it is unable to retrieve an IP address from the device it is connected	Internet	250ms ON/ 250ms OFF
LED-08	Line Registration failed: Device is properly setup, can connect to provider's network, but cannot register to provider's SIP proxy (no 200 OK)	Phone	2x1000ms ON/OFF + 3sec OFF

LED-09	Device is connected (has physical data link) but there are incorrect network settings typically associated with PPPoE connection failure	Internet	250ms ON/ 250ms OFF
LED-10	Hazardous potential test failed: Hazardous AC or DC voltage is present on the tip and ring or both signals of phone line X	Phone	1x250ms ON/OFF + 3sec OFF
LED-11	Foreign electro Motive Force (EMF) Test fail. Foreign voltage is present on the tip, ring or both signals of phone line. Device has detected additional external Phone voltage on the FXS phone line.	Phone	2x250ms ON/OFF + 3sec OFF
LED-12	Resistive fault test failed. Either tip or ring is shorted to ground or they are shorted to each other.	Phone	3x250ms ON/OFF + 3sec OFF
LED-13	Receiver off hook test fail. One or more phones are off hook on phone line during test.	Phone	4x250ms ON/OFF + 3sec OFF
LED-14	REN test failed – high REN detected. Too many parallel phones connected to phone line X	Phone	5x250ms ON/OFF + 3sec OFF
LED-15	Line is active	Phone	500ms ON/OFF
LED-16	Line inactive	Phone	ON
LED-17	During Provisioning Stage*	Internet / Phone	500ms ON/OFF
LED-18	During Firmware Recovery Stage*	Internet / Phone	250ms ON/OFF
LED-19	Line X is registered normally to the sip providers network and is ready to make a call	Phone	2x1000 ms ON/OFF + 3sec OFF

*Note: In Provisioning and Firmware Recovery Stage, the power LED is Steady ON.
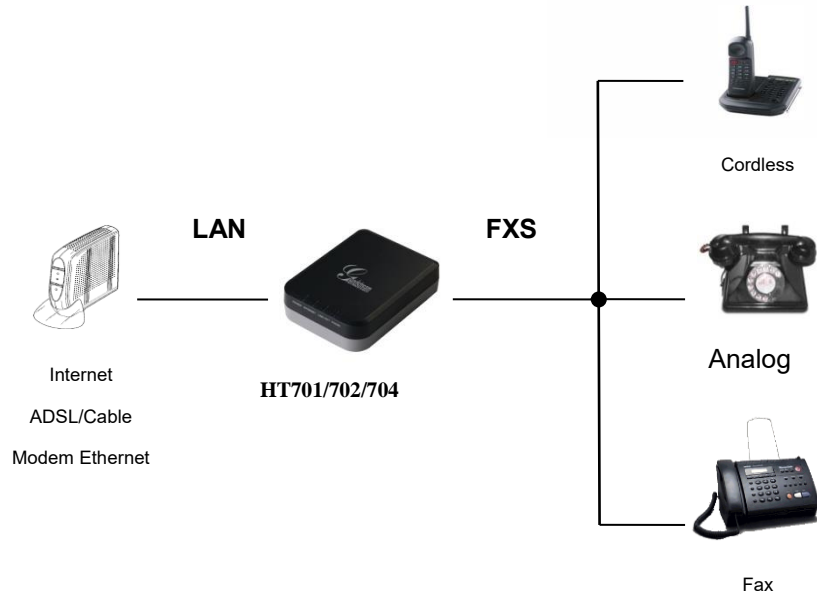
Figure 2: Connection of HT70X

PRODUCT OVERVIEW

The HT70X is a full feature voice and fax-over IP device that offers a high-level of integration including a 10M/100Mbps network port and one FXS telephone port, market-leading sound quality, rich functionalities, and a compact and lightweight design. The VoIP network signaling protocol supported is SIP. The HT70X is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market. Moreover, it supports comprehensive voice codecs including G.711 (a/μ-law), G.723.1, G.726-32, G.729 and iLBC.

Software Features Overview

- Supports Voice Codecs:
- G.711 (a/μ-law), G.723.1, G.726-32, G.729 and iLBC.
- T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation, Voice Activation Detection (VAD),
- Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets

Table 4: HT70X Software Features

	HT 701	HT 702	HT 704
Telephone Interfaces	1 FXS ports	2 FXS ports	4 FXS ports
SIP Provisioning	1 Sip Account, 1 Profile	2 Sip Accounts, 2 Profiles	4 Sip Accounts, 2 Profiles
Number of Concurrent Calls	1 Concurrent Call	2 Concurrent Calls	4 Concurrent Calls
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, Packetized Voice Protocol Unit (supports RTP/RTCP protocol), G.168 compliant Echo Cancellation, LEC (line echo cancellation) with NLP, Asymmetric RTP stream		

Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1, G.726-32(ADPCM), G.729, iLBC, G.726-32 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status
DHCP Server/Client	Yes, DHCP Client only
Telnet Server	
Fax over Ip	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for
QoS	
Transport Protocol	
DTMF Method	Flexible DTMF transmission method, user interface of In-audio, RFC2833, and/or
IP Signaling	
Provisioning	TFTP, HTTP, HTTPS, TR-069, XML
Control	TLS/SIPS , SIP over TCP/TLS
Device Management	Web interface or via secure encrypted AES or non-encrypted central configuration file for mass deployment using Grandstream binary file or xml format. Auto/manual provisioning system or via built-in IVR. NAT-friendly remote software upgrade (via TFTP/HTTP/HTTPS) for deployed devices including behind firewall/NAT. Syslog support. Full support of TR-069 management protocol and Telnet access.
Dial Plan	*Yes*
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Call Handling Features	Caller ID display or block, Call waiting caller ID, Call waiting/flash, Call transfer, hold, forward, 3-way conferencing, message waiting, Do-Not-Disturb (DND), call-return service

Hardware Specification

The table below lists the Hardware and Technical specification of HT70X.

Table 5: HT70X Hardware And Technical Specifications

	HT701	HT702	HT704
Telephone Interfaces	1 RJ11 FXS port	2 RJ11 FXS ports	4 RJ11 FXS ports
Network Interface	1 RJ45 10/100 Mb Base-TX, Full Duplex		TX, Full Duplex port **with connectivity LEDs**
LED Indicators	POWER, INTERNET, LINK/ACTIVITY, PHONE	POWER, INTERNET, LINK/ACTIVITY, PHONE1, PHONE2	POWER, PHONE1, PHONE2, PHONE3, PHONE4
Factory Reset Button	Yes		
Universal Switching Power Adaptor	*Input:* 100–240 VAC/50-60 Hz 0.18A Max *Output:* 12VDC, 0.5A, **UL** certified		*Input:* 100–240 VAC/50-60 Hz 0.18A Max *Output:* 12VDC, 1A, **UL** certified
Environmental	Operational: 32º–104ºF or 0º–40ºC Storage: 14º–140º F or -10 º –60º Humidity: 10–90% Non-condensing		
Dimensions (H x W x D)	26 x 65 x 86mm	28 x 115 x 75mm	28 x 115 x 75mm
Short Haul Loop	5REN, Up to 1Km on 24 AWG wire	3REN: Up to 1Km on 24 AWG line	
Polarity Reversal / Wink	Yes		
EMC	FCC part15 Class B, EN55022, EN55024, CISPR22, and CISPR24		

Safety	EN60950-1 & UL60950-1 (UL only for PSU)
Compliance	FC CE C

BASIC OPERATIONS

Understanding HT70X Interactive Voice Prompt Response Menu

The HT70X has a built-in voice prompt menu for simple device configuration which lists actions, commands, menu choices, and descriptions. The IVR menu work with any phone connected to the HT70X . Pick up the handset and dial "***" to use the IVR menu.

Table 6: Voice Prompt Menu

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*" for the next menu option Press "#" to return to the main menu Enter 01-05, 07,10, 13-17,47 or 99 menu options
01	"DHCP Mode", "Static IP Mode"	Press "9" to toggle the selection If using "Static IP Mode", configure the IP address information using menus 02 to 05. If using "Dynamic IP Mode", all IP address information comes from the DHCP server automatically after reboot.
02	"IP Address " + IP address	The current WAN IP address is announced If using "Static IP Mode", enter 12 digit new IP address. **You need to reboot the HT70X for the new IP address to take Effect.**
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
07	Preferred Vocoder	Press "9" to move to the next selection in the list: • PCM U / PCM A • iLBC • G-726 • G-723 • G-729
10	"MAC Address"	Announces the Mac address of the unit.

13		Announces current Firmware Server IP address. Enter 12 digit new IP address.
		Announces current Config Server Path IP address. Enter 12 digit new IP address.
17	Firmware Upgrade	Firmware upgrade mode. Press "**9**" to toggle among the following three options: • always check • check when pre/suffix changes • never upgrade
47	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone.　(See "*Make a Direct IP Call*".)
86	Voice Mail	Access to your voice mails messages.
99	"RESET"	Press "**9**" to reboot the device Enter MAC address to restore factory default setting (See **Restore Factory Default Setting** section)
	"Invalid Entry"	Automatically returns to main menu
	"Device not registered"	This prompt will be played immediately after off hook If the device is not register and the option "Outgoing Call without Registration" is in NO

Notes:

- "*" shifts down to the next menu option and "**#**" returns to the main menu.
- "9" functions as the ENTER key in many cases to confirm or toggle an option.
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168000026. No decimal is needed).
- Key entry cannot be deleted but the phone may prompt error once it is detected.

Placing a phone call

Phone or Extension Numbers

To make the outgoing calls using your HT70X :

1. Pick up the handset of the connected phone;
2. Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"); or
3. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial an extension directly on the same proxy, (e.g. 1008), and then press the # or wait for 4 seconds;
2. Dial an outside number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

Direct IP Calls

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

Elements necessary to completing a Direct IP Call:

- Both HT70X and other VoIP Device, have public IP addresses, or
- Both HT70X and other VoIP Device are on the same LAN using private IP addresses, or
- Both HT70X and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

The HT70X supports two ways to make Direct IP Calling:

Using IVR

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

Using Star Code

1. Pick up the analog phone then dial "*47";
2. Enter the target IP address.

Note: No dial tone will be played between step 1 and 2 and destination ports can be specified using "*" (encoding for ":") followed by the port number.

Examples of Direct IP Calls:

a) If the target IP address is 192.168.0.160, the dialing convention **is *47 or Voice Prompt with option 47, then 192*168*0*160**, followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified;

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: ***47 or Voice Prompt with option 47, then 192*168*0*160*5062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

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

Call Hold

You can place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button).

Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

Call Waiting

The call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled.

To toggle between incoming call and current call, you need to press the "flash" button the first call is placed on hold.

Press the "flash" button to toggle between the active calls.

Call Transfer

Blind Transfer

Assume that the call is established between phone A and B are in conversation. The phone A wants to *blind transfer* phone B to phone C:

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

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

Attended Transfer

Assume that the call is established between phone A and B are in conversation. The phone A wants to *attend transfer* phone B to phone C:

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

Note: When attended transfer fails and A hangs up, the HT70X will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-Way Conferencing

The HT70X supports Bellcore style 3-way Conference. To perform the 3-way conference, we assume that the call is established between phone A and B are in conversation. Phone A(HT70X) wants to bring third phone C into conference:

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

6. If phone A hangs up, the conference will be terminated for all three parties when configuration "Transfer on Conference Hang up" is set to "No". If the configuration is set to "Yes", A will transfer B to C so that B and C can continue the conversation.

Call Return

To call back to the latest incoming number.

1. Pick up the handset of the connected phone (Off-hook).
2. After hearing the dial tone, input "*69", and follow by "#" or wait for time out.
3. Your phone will automatically call back to the latest incoming number.

Note: All star code (*XX) related features mentioned above are supported by ATA default settings. If your service provider provides different feature codes, please contact them for instructions.

CALL FEATURES

The HT70X support all the traditional and advanced telephony features.

Table 7: HT70X Call Features

Key	Call features
	Forcing a Codec (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *027201 (iLBC).
	Disable LEC (per call) Dial "*03" +" number". **No** dial tone is played in the middle.
	Enable SRTP.
	Disable SRTP.
	Block Caller ID (for all subsequent calls).
	Send Caller ID (for all subsequent calls).
	Direct IP Calling. Dial "*47" + "IP address". **No** dial tone is played in the middle.
	Disable Call Waiting (for all subsequent calls).
	Enable Call Waiting (for all subsequent calls).
	Block Caller ID (per call). Dial "*67" +" number". **No** dial tone is played in the middle.
	Send Caller ID (per call). Dial "*67" +" number". **No** dial tone is played in the middle.
	Call Return Service: Dial *69 and the phone will dial the last incoming phone number received.
	Disable Call Waiting (per call). Dial "*70" +" number". **No** dial tone is played in the middle.
	Enable Call Waiting (per call). Dial "*71" +" number". **No** dial tone is played in the middle.

	Cancel Busy Call Forward. To cancel "Busy Call Forward", dial "*91", wait for dial tone, then
	Pressing pound sign will serve as Re-Dial key.

CONFIGURATION GUIDE

Configuring HT70X through Voice Prompts

DHCP MODE

Follow Table 4 with voice menu option 01 to enable HT70X to use DHCP.

STATIC IP MODE

Follow Table 4 with voice menu option 01 to enable HT70X to use STATIC IP mode, then use option 02, 03, 04 to set up HT70X's IP, Subnet Mask, Gateway respectively.

FIRMWARE SERVER IP ADDRESS

Select voice menu option 13 to configure the IP address of the firmware server.

CONFIGURATION SERVER IP ADDRESS

Select voice menu option 14 to configure the IP address of the configuration server.

UPGRADE PROTOCOL

Select voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP, HTTP and HTTPS.

FIRMWARE UPGRADE MODE

Select _voice menu option 17_ to choose firmware upgrade mode.   There are three options:
1) always check, 2) check only when pre/suffix changes, and 3) never upgrade

Configuration via Web Browser

The HT70X embedded Web server responds to HTTP GET/POST requests. Embedded HTML pages allow a user to configure the HT70X through a web browser such as Google Chrome, Mozilla Firefox and Microsoft's IE.

Accessing the Web UI

1. Connect the computer to the same network as your HT70X.

2. Make sure the HT70X is booted up.

3. You may check your HT70X IP address using the IVR on the connected phone.

4. Please see Obtain HT70X IP Address Via Connected Analogue Phone.

5. Open Web browser on your computer.

6. Enter the HT70X's IP address in the address bar of the browser.

7. Enter the administrator's password to access the Web Configuration Menu.

Notes:

- The computer must be connected to the same sub-network as the HT70X . This can be easily done by connecting the computer to the same hub or switch as the HT70X .

- Recommended Web browsers:
    - **Microsoft Internet Explorer:** version 10 or higher.
    - **Google Chrome**: version 58.0.3 or higher.
    - **Mozilla Firefox**: version 53.0.2 or higher.
    - **Safari**: version 5.1.4 or higher.
    - **Opera**: version 44.0.2 or higher.

Web UI Access Level Management

There are two default passwords for the login page:

User Level	Password	Web Pages Allowed
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	All pages

The password is case sensitive with maximum length of 25 characters.

When changing any settings, always submit them by pressing the **Update** or **Apply** button on the bottom of the page. After submitting the changes in all the Web GUI pages, reboot HT70X to have the changes take effect if necessary; most of the options under the **Advanced Settings** and **FXS Port (x)** pages require reboot.

Saving the Configuration Changes

After users makes changes to the configuration, pressing the **Update** button will save but not apply the changes until the **Apply** button is clicked. Users can instead directly press the **Apply** button. We recommend rebooting or powering cycle the phone after applying all the changes.

Changing Admin Level Password

1. Access your HT70X web UI by entering its IP address in your favorite browser (screenshots below are from HT70X but the same applies to HT70X 1).
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings and navigate to **Advanced Settings > Admin Password.**
4. Enter the new admin password.
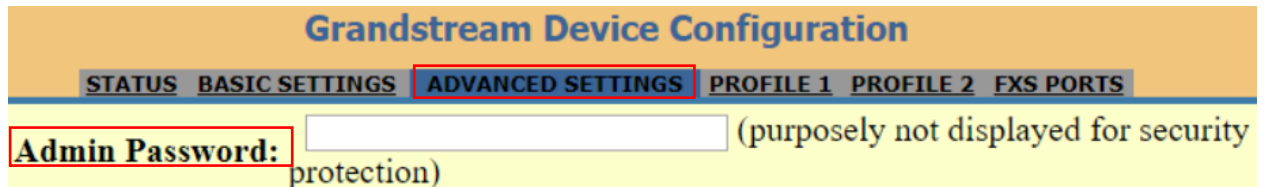5. Press **Apply** at the bottom of the page to save your new settings.

Figure 3: Admin Level Password

Changing User Level Password

1. Access your HT70X web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings and navigate to **Basic Settings > End User Password.**
4. Press **Apply** at the bottom of the page to save your new settings.
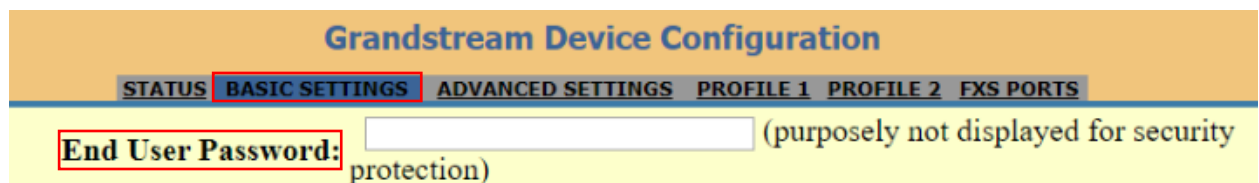
Figure 4: User Level Password

Changing HTTP Web Port

1. Access your HT70X web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings and navigate to **Basic Settings > Web Port.**
4. Change the current port to your desired/new HTTP port. Ports accepted are in range [1-65535].
5. Press **Apply** at the bottom of the page to save your new settings.

NAT Settings

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

1. STUN Server (under advanced settings webpage) Enter a STUN server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
2. Use Random SIP/RTP Ports (under advanced settings webpage) This setting depends on your network settings. Generally, if you have multiple IP devices under the same network, it should be set to Yes. If using a public IP address, set this parameter to No.
3. NAT traversal (under the FXS web page) Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

The HT70X support the following DTMF mode:

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

Set priority of DTMF methods according to your preference. This setting should be based on your server DTMF setting.

Preferred Vocoder (Codec)

The HT70X supports following voice codecs. On FXS ports pages, choose the order of your favorite codecs:

- PCMU/A (or G711µ/a)
- G729 A/B
- G723.1
- G726
- iLBC

Configuring the HT70X through Voice Prompts

As mentioned previously, The HT70X has a built-in voice prompt menu for simple device configuration. Please refer to "Understanding HT70X Interactive Voice Prompt Response Menu" for more information about IVR and how to access its menu.

- **DHCP MODE**

Select voice menu option 01 to allow the HT70X to use DHCP.

- **STATIC IP MODE**

Select voice menu option 01 to allow the HT70X to enable the STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

- **FIRMWARE SERVER IP ADDRESS**

Select voice menu option 13 to configure the IP address of the firmware server.

- **CONFIGURATION SERVER IP ADDRESS**

Select voice menu option 14 to configure the IP address of the configuration server.

- **UPGRADE PROTOCOL**

Select the menu option 15 to choose firmware and configuration upgrade protocol between TFTP, HTTP and HTTPS.

- **FIRMWARE UPGRADE MODE**

Select voice menu option 17 to choose firmware upgrade mode among the following three options: "Always check, check when pre/suffix changes, and never upgrade".

Table 8: Basic Settings

Telnet Server	Default is set to **Yes**.
HTTP Access	Default is set to **Yes**. If set to No, http access will be denied.

IP Address	There are two modes to operate the HT70X:
	DHCP mode: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HT701 acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The HT70x will establish a PPPoE session if any of the PPPoE fields is set.
	Static IP mode: configure the IP address, Subnet Mask, Default Router IP address, 1st Preferred DNS Server, 2nd Preferred DNS Server, 3rd Preferred DNS Server, 4th Preferred DNS Server. These fields are set to zero by default.
DHCP hostname	Default is **blank**. This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
DHCP domain	Default is **blank**. This option specifies the domain name that client should use when resolving hostnames via the Domain Name System.
PPPoE account ID	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	Default is **blank**. This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here.
Preferred DNS server	The preferred DNS Server to be used
Time Zone	Controls how the date/time is displayed according to the specified time zone.

Self-Defined Time Zone	The syntax is **std offset dst [offset],start[/time],end[/time]** Default is set to : **MTZ+6MDT+5,M3.2.0,M11.1.0** **MTZ+6MDT+5,** Time zone with 6 hours offset with 1 hour ahead which is the US central time. It is positive (+) if the local time zone is west of the Prime Meridian and negative (-) if it is east. **Prime Meridian (a.k.a: International or Greenwich Meridian)** **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 etc) The 3rd number indicates Weekday: 0,1, 2, ..,6(for Sun, Mon, Tue, .., Sat) Therefore, this example is the DST which starts from the second Sunday of March to the 1st Sunday of November.
Allow DHCP server to set Time Zone	Default **Yes**. Let the DHCP server handle the Time Zone
Language	Languages supported with voice prompt and web interface, except Spanish that it is only in IVR.
Reset Type	Gives the user the option to set to default all VoIP related configuration (mainly everything located on FXS/Profile page), all ISP (Internet Service Provider) configuration which may affect the IP address, or both at the same time. Note: After you choose the reset type, you will have to click the reset button for it to take effect.

In addition to the Basic Settings configuration page, end users also have access to the Device Status page.

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting. The Mac address will appear in this place. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the bottom panel of the device.
Product Model	This field contains the product model info.
Hardware Version	This field shows the hardware revision of the unit and the part number.
Software Version	*Program:* This is the main software release. *This number is always used for firmware upgrade.* Current release is 1.0.9.4. Boot and Loader are seldom changed. *Bootloader:* current version is 1.0.0.6. *Core:* current version 1.0.9.4. *Base:* current version is 1.0.9.4. *CPE:* gs_cpe version number information.
PPPoE Link Up	Indicates whether the PPPoE connection is up if the HT70X is connected to DSL modem.
NAT	This filed indicates the type of NAT connection used by the HT70X.

Port Status							

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS	On Hook	Registered	Yes	613		

Log in to the advanced user configuration page the same way as for the basic configuration page.   The password is case sensitive and the factory default password for Advanced User is "*admin*".

Advanced User configuration includes the end user configuration and the advanced configurations including: a) SIP configuration, b) Codec selection, c) NAT Traversal Setting and d) other miscellaneous configuration. HT701 and HT702 FXS SIP account(s) have its own configuration page. HT704 has two (2) profiles for the four (4) FXS ports.

Table 10: Advanced Settings

Layer 2 QoS	
STUN Server is	
Keep-alive interval	This parameter specifies how often the HT70X sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.

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Use STUN to detect network connectivity	Use STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times, the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled. Default setting is **No.**
Firmware Upgrade and Provisioning	Enables HT70X to download firmware or configuration file through either the TFTP, HTTP or HTTPS server.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the HT70X retrieves the new configuration file or new code image from the specified TFTP server at boot time. After 5 attempts, the system will timeout and will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image is saved into the Flash memory. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the local network, this process can take up to 15 or 20 minutes.
Via HTTP / HTTPS Server	The URL for the HTTP/HTTPS server used for firmware upgrade and configuration via HTTP. *For example*, http://provisioning.mycompany.com:6688/Grandstream/1.0.0.67"**:6688**" is the specific TCP port where the HTTP or HTTPS server is listening; it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, HT70X will only do HTTP/HTTPS download once at boot up.
Firmware Server Path	
Config Server Path	IP address or domain name of configuration server.
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.

	The user name needed to authenticate with the HTTP/HTTPS server.
HTTP/HTTPS Password	The password needed to authenticate with the HTTP/HTTPS server.
Always send HTTP Basic Authentication Information	Default is Send **HTTP Basic Authentication Information only when challenged**. If set to **Always**, device will send configured user name and password within HTTP request without server sending authentication challenge.
Firmware File Prefix	Default is **blank**. If configured, HT701 will request firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
Firmware File Postfix	Default is **blank**. End user should keep it blank.
Config File Prefix	Default is **blank**. End user should keep it blank.
Config File Postfix	Default is **blank**. End user should keep it blank.
Allow DHCP Option 66 or 160 to override server	If set to "Yes", configuration and upgrade server information can be obtained using DHCP option 66 or option 160 from DHCP server located in customer's environment. Default setting is **Yes.**
3CX Auto Provision	Enables automatic provision feature on the phone when 3CX is used as the SIP server. The default setting is "Yes".
Automatic Upgrade	Choose "*Yes*" to enable automatic upgrade and provisioning. If select "Check every minutes" input the amount of minutes you want it to check for update. If select "Yes, daily at hour" make sure to input the hour of the day you want it to check for update, e.g. for 11 pm type 23. If select "Yes, weekly on day" make sure you input the day of the week (in format 0-6, 0 is Sunday) you want it to check for update. When set to No, HT70X will only do the following option you select; "Always check for New Firmware at Boot up" will check for new firmware every time the device reboots. "Check New Firmware only when F/W pre/suffix changes" will check for updates only when the pre/suffix has been changed.

	If set to Yes, config file is authenticated before acceptance. This protects the configuration from an unauthorized change.
SIP TLS Certificate	The user specify SSL certificate used for SIP over TLS in X.509 format.
SIP TLS Private Key	The user specify SSL private key used for SIP over TLS in X.509 format.
SIP TLS Private Key Password	User specify password to protect the private key above.
ACS URL	User specify the Auto Configuration Server's URL (TR-069 protocol)
ACS Username	User specify the ACS Username
ACS Password	User specify the ACS password
Periodic Inform Enable	Default is **No**. If set to YES, device will send inform packets to the ACS
Periodic Inform Interval	Frequency that the inform packets will be sent out to the ACS
Connection Request Username	Set a user name for the ACS to connect to this device
Connection Request Password	Set a password for the ACS to connect to this device
CPE SSL Certificate	The Cert File for the phone to connect to the ACS via SSL.
CPE SSL Private Key	The Cert Key for the phone to connect to the ACS via SSL.
Country Specific Deployment	Default is **None**. If choose any other option, FXS related settings will be applied to most commonly used value in the configured the region.
System Ring Cadence	Configuration option is set ring cadence on all FXS ports for all incoming calls. (Syntax: c=on1/off1-on2/off2-on3/off3(only 3 cadences maximum)) Default is set to **c=2000/4000;** (US standards)

Call Progress Tones	Using these settings, users can configure tone frequencies and cadence according to their preference. By default they are set to North American frequencies. Configure these settings with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Example configuration for N.A. Dialtone: **f1=350@-13,f2=440@-13,c=0/0;** **Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]** **(Note: freq: 0 - 4000Hz; vol: -30 - 0dBm)**
Disable Voice Prompt	Default is **No**. Disables the voice prompt configuration.
Disable Direct IP Call	Default is **No**. Disables the Direct IP Call function.
Disable Attended Transfer	Default is **No**. Attended transfers will be disabled if set to **Yes**
NTP server	URI or IP address of the NTP (Network Time Protocol) server. This parameter synchronizes the date and time.
Allow DHCP option 42 to override NTP serve	Default **Yes.** Enables the DHCP server to handle the NTP Server via Option 42

	The IP address or URL of System log server. This feature is especially useful for
Send SIP Log	Default is **No.** If Enabled, the HT70X will send a replicate of the SIP packets on the syslog.

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Primari RADIUS Acct Port	Default is **1813.** Specifies the port to be used for the Primary RADIUS Account. **HT704 Only**
Primary RADIUS Server Secret	Specifies the secret string to be used to authenticate the RADIUS connection to the Primary Server. It should match RADIUS configuration. **HT704 Only**
Secondary RADIUS Server	Set the IP or FQDN of the Secondary RADIUS Server. **HT704 Only** In case Primary Radius server becomes unusable, secondary will take role of primary and will manage credit recourses in the network
Secondary RADIUS Auth Port	
Secondary RADIUS Acct Port	Default is **1813.** Specifies the port to be used for the Secondary RADIUS Account. **HT704 Only**
RADIUS Retry	Default is **3. HT704 Only**

Table 11: Account Settings

Primary SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Failover SIP Server	Failover SIP Server's IP address or Domain name in case primary server does not respond.
Prefer Primary SIP Server	Default is **No**. If set to yes, it will register to Primary Server if registration with Failover server expires
	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by HT70X for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
	Default is **No**. If set to Yes, device will collect SIP server address from DHCP option 120 for FXS1 (HT701/2) or Profile1 (HT704).
SIP transport	User can select UDP or TCP or TLS. Default is **UDP.**
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number. **HT701 and HT702 only**
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID. **HT701 and HT702 only**
Authenticate Password	SIP service subscriber's account password. **HT701 and HT702 only**
Name	SIP service subscriber's name for Caller ID display. **HT701 and HT702 only**

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NAT Traversal (STUN)	This parameter defines whether or not the HT70X NAT traversal mechanism is activated. If activated (by choosing "Yes") and a STUN server is also specified, then the HT70X performs according to the STUN client specification. Using this mode, the embedded STUN client will detect if and what type of firewall/NAT. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HT70X will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the HT70X 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.
DNS Mode	One from the 3 modes are available for "DNS Mode" configuration: -A Record (for resolving IP Address of target according to domain name) -SRV (DNS SRV resource records indicates how to find services for various protocols) -NAPTR/SRV (Naming Authority Pointer according to RFC 2915) One mode can be chosen for the client to look up server. The default value is "A Record"
Tel URI	The default setting is **"Disabled".** If the phone has an assigned PSTN Number, this field should be set to "User=Phone" then a "User=Phone" parameter will be attached to the "From header" in the SIP request to indicate the E.164 number. If server supports TEL URI format, then this option needs to be selected.
SIP Registration	Controls whether the HT701 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.
Outgoing Call without Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by Internet Telephone Service Provider) but is unable to receive incoming calls.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HT70X 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).

Reregister before Expiration	Default is **0** (function disabled). If configured with none 0 value. Then this parameter will specify the time that HT70X refreshes its registration before expired, instead of the default refresh time interval before expiration, which is half of the Register Expiration time or 10min maximum.
SIP Registration Failure Retry Wait Time	Retry registration if the process failed. Default is **20** seconds.
SIP Registration Failure Retry Wait Time upon 403 Forbidden	Retry registration if the process failed upon 403 forbidden. Default is 1200, valid value is between 0-3600. 0 means stop retry registration upon 403 response.
Enable SIP OPTIONS Keep Alive	Default is **No**. If set to Yes, it will use SIP OPTIONS instead of STUN as keep alive messages.
SIP OPTIONS Keep Alive Interval	Configure the Interval time of SIP OPTIONS keep-alive messages. Between 1 - 64800 seconds. Default is **30**.
SIP OPTIONS Keep Alive Max Lost	Configure the number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3 - 10 packets, default is **3**.
Layer 3 QoS	Input value for SIP DSCP (Diff-Serv value in decimal, default 24) And value for RTP DSCP (Diff-Serv value in decimal, default 46)
Local SIP port	Defines the local SIP port the HT70X will listen and transmit. The default value for FXS port is **5060**.
Local RTP port	Defines the local RTP-RTCP port pair the HT70X will listen and transmit. It is the base RTP port for channel 0. When configured, **channel 0** uses this port _value for RTP and the port_value+1 for its RTCP The default value for FXS port is **5004**.
Use Random SIP Port	Default is **No**. This parameter forces the random generation of The local SIP ports when set to Yes. This is usually necessary when multiple HT70X are behind the same NAT.
Use Random RTP Port	Default is **No**. This parameter forces the random generation of the local RTP ports when set to Yes. This is usually necessary when multiple HT70X are behind the same NAT.

Hold Target Before Refer	Default is **Yes**. Allows user to hold the phone call before refer it. If set to **No**, the call will not be hold before referred.
Refer to Use Target Contact	Default is **No**. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
Transfer on Conference Hang up	Default is **No**. In which case if the conference originator hangs up the conference will be terminated. When option YES is chosen, originator will transfer other parties to each other so that B and C can choose to either continue the conversation or hang up.
Enable Ring-Transfer	Default is **No**, this will create a Semi-Attendant Transfer. When set to Yes, device can transfer the call upon receiving ring back tone or SIP message 180.
Disable Bellcore Style 3-Way Conference	Default is **No**. you can make a Conference by pressing 'Flash' key. If set to **Yes**, you need to dial *23 + second callee number.
Remove OBP from Route Header	Default is **No**. When option YES is chosen, the Out Bound Proxy will be removed from Route header.
Support SIP Instance ID	Default is **Yes**. If set to Yes, the contact header in REGISTER request will contain SIP Instance ID as defined in IETF SIP Outbound draft.
Validate incoming SIP message	Default is **No**. If set to yes all incoming SIP messages will be strictly validated according to RFC rules. If message will not pass validation process, call will be rejected.
Check SIP User ID for incoming INVITE	Default is **No.** Check the incoming SIP User ID in Request URI. If they don't match, the call will be rejected. If this option is enabled, the device will not be able to make direct IP calls.
Authenticate incoming INVITE	Default is **No.** If set to Yes, device will challenge the incoming INVITE for the Authenticate ID and Password with 401 Unauthorized.
Authenticate server certificate domain	Default is **No.** If set to Yes, device will authenticate server certificate domain.
Authenticate server certificate chain	Default is No. If set to Yes, device will authenticate server certificate chain.
Trusted CA certificates	Default is null. Or you can configure your trusted CA certificates manual after enabled Authenticate server certificate domain.

Allow Incoming SIP Messages from SIP Proxy Only	Default is **No.** Check the incoming SIP messages. If they don't come from the SIP proxy, they will be rejected. If this option is enabled, the device will not be able to make direct IP calls.
Use Privacy Header	If set to **Default**, it will add Privacy header unless special feature is **Telkom SA** or **CBCOM.**
Use P-Preferred-Identity Header	If set to **Default**, it will add P-Preferred-Identity header unless special feature is **Telkom SA** or **CBCOM.**
SIP REGISTER Contact Header Uses	Default is **LAN Address**. If set to **WAN Address**, device will detect its WAN address and use it in SIP REGISTER Contact Header.
SIP T1 Timeout	T1 is an estimate of the round-trip time between the client and server transactions. If the network latency is high, select larger value for more reliable usage. Default is **0.5 Sec.**
SIP T2 Interval	Maximum retransmission interval for non-INVITE requests and INVITE responses. Default is **4 Sec.**
SIP Timer D	Configure the SIP Timer D defined in RFC3261. 0 - 64 seconds. Default 0
DTMF Payload Type	Sets the payload type for DTMF using RFC2833. Default is **101.**
Preferred DTMF method	The HT70X supports up to 3 different DTMF methods including in-audio, via RTP (RFC2833) and via Sip Info using SIP INFO messages. The user can configure DTMF method in a priority list.
Disable DTMF Negotiation	Default is **No**. If set to yes, use above DTMF order without negotiation
Generate Continuous RFC2833 Events	Default is **No.** If set to yes, RFC2833 events are generated until key is released.
Send Hook Flash Event	Default is **No.** If set to yes, flash will be sent as DTMF event.
Flash Digit Control	Default is **No**. If set to Yes will overrides the default settings for call control when both channels are in use.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number that is automatically dialed when off-hook. <u>Only the user part</u> of a SIP address needs to be entered here. The HT70X will automatically append the "@" and the host portion of the corresponding SIP address. **HT701 and HT702 only.**

	The auto-dial delay after off hook.
	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Disable Multiple m line in SDP	Default is **No**. If set to Yes, device will send only one m line in SDP, regardless how many m field in the incoming SDP.
Distinctive Ring Tone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls using the FXS port or Profile. Distinctive ring tones can be configured not only for matching a whole number, but also for matching prefixes. In this case symbol * (star) will be used. **For example:** if configured as *617, Ring Tone 1 will be used in case of call arrived from the area code 617. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page. **Note:** If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. Bellcore rings and tones are independent from custom ring tones. The custom ring tones can also be specified by alert-info header, for example: Alert-Info: <http://127.0.0.1>;info=ring5
Disable Call Waiting	Default is **No**. If set to Yes Call Waiting indication information will not be provided to analog phone connected to this FXS port.
Disable Call-Waiting Caller ID	Default is **No**. If set to Yes Call Waiting caller ID will not be provided to analog phone connected to this FXS port.

Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting information arrives. The CWCID information will still be displayed.
Disable Connected Line ID	Default is No. If set to Yes will disable the connect line ID appear on device.
Disable Receiver Offhook Tone	Default is **No.** If set to Yes, it will disable the warning to alert that the phone has been left off-hook for an extended period of time.
Disable Reminder Ring for On-Hold Call	Default is **No**. Do not play the reminder ring when this is set to Yes.
Disable Visual MWI	Default is **No.** If set to "Yes", the MWI information will not be transferred to the analog phone connected to the FXS port.
Ring Timeout	Default value is **60 Sec.** Incoming call will stop ringing when not picked up given a specific period of time.
Hunting Group Ring Timeout	Default value is **20 Sec.** If call is not answered within this designated time period, the call will be forwarded to the next member of a Hunt Group. **HT704 only.**
Hunting Group Type	Linear and Circular. Default is **Circular.** Linear style will sort the call to the lowest-numbered available line, this is also called "serial hunting". Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the firstline. Lines are skipped if they are still busy on a previous call. **HT704 only**
Delayed Call Forward Wait Time	Default value is **20 seconds**. In case this feature activated using * codes (*92 code), the call will be forwarded after this preconfigured amount of time.
No Key Entry Timeout	Default is **4 seconds**. Dialing process is completed and outgoing call is initiated if no key entry occurs during this preconfigured interval.
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response.

	Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature does NOT work with and should NOT be enabled for direct IP-to-IP calling.*
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Default is **Yes.** It allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", "#" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "#" key can be included as part of number.

Dial Plan	**Dial Plan Rules**: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; - xx+ - at least 2 digits number; xx. – at least 2 digit number. ^ - exclude; [3-5] - any digit of 3, 4, or 5; [147] - any digit 1, 4, or 7; <2=011> - replace digit 2 with 011 when dialing < =1> - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed | - or **Example 1:** {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 11 digit numbers with leading digits 1617 **Example 2:** {^1900x+ | <=1617>xxxxxxx} – Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} –Allow any combinations of numbers with 11 digits which has a leading digit 1, but 5th digit cannot be 0 or 1. Or any length of numbers with a minimum of 2 digits beginning with 2, with the leading digit replaced with 011. 1. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US:

	{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }
	Explanation of example rule (reading from left to right):
	• **^1900x. -** prevents dialing any number started with 1900
	• **<=1617>[2-9]xxxxxx -** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically
	• **1[2-9]xx[2-9]xxxxxx |-** allows dialing to any US/Canada Number with 11 digits length
	• **011[2-9]x. -** allows international calls starting with 011
	• **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911
	Note: In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan should be: **{ *x+ }**.
Subscribe for MWI	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	Default is **No**. If this parameter is set to "Yes", the "From" header along with Privacy and P_ Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors.
Session Expiration	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. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is **180 seconds**.
Min-SE	The minimum session expiration (in seconds). The default value is **90 seconds**.
Caller Request Timer	Default is **No**. If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	Default is **No**. If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.

Force Timer	Default is **No**. 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	Default is **Omit.** 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	Default is **UAC.** 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. Default is **No**,
Send Re-INVITE After Fax	Default is **No**, If set to "Yes", device will send an INVITE with audio vocoders upon completion of Fax to continue session in audio only.
Enable Silence Detection for Fax Disconnect	For fax machines that do not send a Disconnect when fax is done. This option Enables/Disables the detection of silence in order to know the fax has finished. The silence period is non-configurable and fixed to 7 seconds. Default is **No**,
Enable 100rel	Default is **No**, If set to Yes, Enables the use of PRACK (Provisional Acknowledgment) method.
Add Auth Header On Initial REGISTER	If set to **Yes**, device will include authorization header in the Register request.
Conference URI	Default is null, this option allows to manually configure conference URL.
Use First Matching Vocoder in 200OK SDP	Default is **No**. If set to "Yes", device will include only the first match vocoder in its 200OK response, otherwise it will include all match vocoders in same order received in INVITE.
Preferred Vocoder	The HT70X supports up to 5 different Vocoder types including G.711 A-/U-law, G.726-32, G.723.1, G.729A/B/E, iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8".

Vocoder types can also be changed per call basis by using a star code. Please see the "Call features" section. |
| **Voice Frames per TX** | Default is **2**. Defines the number of voice frames sent in each packet. |
| **G723 Rate** | Default is **6.3kbps**. Defines the encoding rate for G.723.1 vocoder. |

	Default is **20ms**. Sets the iLBC frame size in 20ms or 30ms.
iLBC Payload type	Defines payload type for iLBC. Default value is **97**. The valid range is between 96 and 127.
VAD	
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Re-Invite after Fax Tone Detection Mode	
Jitter Buffer Type	
Jitter Buffer Length	Select Low, Medium or High based on network conditions. Default is **Medium.** • High (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement • Medium (initial 100ms, min 20ms, max 200ms) • Low (initial 50ms, min 10ms, max 100ms)
SRTP Mode	This option defines different implementation of support SRTP (squired RTP) transmission mode. Select between Disabled, Enabled but not Forced, and Enabled and Forced. Default is **Disabled.**

Loop Current Disconnect	Default is **No**. Set it to Yes if the traditional PBX you are using with HT70X uses this method for signaling call termination. Method initiates short voltage drop on the line when remote (VoIP) side disconnects an active call.
Loop Current Disconnect Duration	Default value is **200.** Here can be configured duration of such voltage drop described in topic above. HT70X supports a Duration Range from 100 to 10000 ms.
	Default is **No**. If set to Yes will enable the analog phone use pulse dialing if it support so.
Hook Flash Timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value. Default values are **300 minimum and 1100 maximum.** HT70X supports a range from 40 to 2000 ms.
On Hook Timing	On-hook timing is the minimum time for an on-hook event to be validated. Default value is **400**. HT70X supports a range from 40 to 2000 ms.
Gain	Voice path volume adjustment. • Rx is a gain level for signals <u>transmitted</u> by FXS • Tx is a gain level for signals <u>received</u> by FXS. Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the Tx Gain Level parameter located on the FXS Port Configuration page. If call volume is too low when using the FXS port (ie. the ATA is at user site), adjust volume using the <u>Rx Gain Level</u> parameter under the FXS Port Configuration page. If voice volume is too low at the other end, user may increase the far end volume using the <u>Tx Gain Level</u> parameter under the FXS Port Configuration page.
Disable Line Echo Canceller (LEC)	Default is **No.** If set to "Yes" LEC will be disabled per call base. Recommended for FAX/Data calls.

	Default is **No.** If set to "Yes" NEC will be disabled per call base. Recommended for FAX/Data calls.
	Configure the time limitation of outgoing calls. 0-180 minutes, default is 0 (No Limit).
	Configure ringing frequency for your phone. 15-60 Hz, default is 20 Hz.
Enable High Ring Power	Default is **No**. If it has difficulties to ring the FAX machine or phone. Enable to push higher ring power through FXS port.
Ring Tones	This function lets you configure ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.
SIP Use ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for HT704 to register to (SIP) servers of ITSP.
Name	Any name to identify this specific user.
Profile ID	Select the corresponding Profile ID between Profile 1 and Profile 2.
Hunting Group	This feature enables the HT704 to register all existing FXS ports with the same phone number. Each incoming call will be routed to first available port in Linear or Circular mode. User may configure all ports as members of the same Hunting Group or it may configure different port combinations for more than one Hunting Group. For example: Ports 1, 3 and 5 are members of the same Hunting Group, the rest of the ports may have separate numbers and may be reached independently. Any port, member of a Hunting Group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port.

	For example: Port 1, 2, and 3 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 2, and 3 are not registered. Ports 2 and 3 will be able to place outbound calls using the SIP account of port 1. Select appropriate value for Hunting Group feature. The original SIP account should be set to Active while the group members should be set to the port number of the Active Port. Example configuration of a Hunting group: FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "Active" FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "None" Hunting Group 1 contains ports 1, 2, 3. FXS port 4 is registered but it is not added to the Hunting Group 1.
Enable Ports	Set **No** to disable FXS port.
Offhook Auto-dial	This feature allows you to automatically dial the number specified in this field as soon as the port is offhooked, i.e. when the receiver on the phone connected to Port# is picked up.
Map to FXP Port #	This is used only when peering with a Grandstream GXW410x. Default is 1, Supported values are 1-8, meaning line 1 to line 8 of the GXW410x device where the port will be mapped to.
Map to FXO Gateway IP	This is used when peering with an FXO gateway of any brand. You have to specifically mention the IP address where the call will be sent to.
and Port	Sip port that will be annexed to the IP address above.

Register a SIP Account

The HT701 supports 1 FXS port which can be configured with 1 SIP account, while HT702 supports 2 SIP accounts and HT704 up to 4 SIP account. Please refer to the following steps to register your accounts via

web user interface.

1. Access your HT701/HT702 web UI by entering its IP address in your favorite browser.

2. Enter your admin password (default: admin) and press **Login** to access your settings.

3. Go to **FXS Port (1** or **2)** pages.

4. In **FXS Port** tab, set the following:

   a. **Account Active** to **Yes.**

   b. **Primary SIP Server** field with your SIP server IP address or FQDN.

   c. **Failover SIP Server** with your Failover SIP Server IP address or FQDN. Leave empty if not available.

   d. **Prefer Primary SIP Server** to **No** or **Yes** depending on your configuration. Set to **No** if no Failover SIP Server is defined. If "**Yes**", account will register to Primary SIP Server when failover registration expires.

   e. **Outbound Proxy:** Set your Outbound Proxy IP Address or FQDN. Leave empty if not available.

      f. **SIP User ID**: User account information, provided by VoIP service provider (ITSP). Usually in the form of digit like phone number or a phone number.

      g. **Authenticate ID**: SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.

      h. **Authenticate Password**: SIP service subscriber's account password to register to SIP server of ITSP. For security reasons, the password will field will be shown as empty.

      i. **Name**: Any name to identify this specific user.

5. Press **Apply** at the bottom of the page to save your configuration.

Figure 5: SIP Accounts Settings

After applying your configuration, your account will register to your SIP Server, you can verify if it has been correctly registered with your SIP server from your HT70X web interface under **Status > Port Status > Registration** (If it displays **Registered**, it means that your account is fully registered, otherwise it will display **Not Registered** so in this case you must double check the settings or contact your provider).

Figure 6: Account Status

Rebooting the HT70X from Remote

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

SOFTWARE UPGRADE

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

Firmware Upgrade through TFTP/HTTP/HTTPS

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

> **e.g.** firmware.mycompany.com:6688/Grandstream/1.0.5.2
>
> **e.g.** firmware.grandstream.com

NOTES:

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

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

- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HT70X will attempt to retrieve the new image files by downloading them into the HT70X's SRAM. During this stage, the HT70X's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HT70X will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.

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

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

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

Note: Alternatively, the user can upload the firmware single file from his/her computer using a local directory. For more instructions see below. Also the user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://support.solarwinds.net/updates/New-customerFree.cfm.   Our latest official release can be downloaded from http://www.grandstream.com/support/firmware .

Instructions for local firmware upgrade:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the HT70X device in the same LAN segment.
3. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

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

Configuration File Download

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

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

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

For more details on Grandstream Device provisioning, please refer to:
http://www.grandstream.com/sites/default/files/Resources/gs-provisioning_guide_public.pdf

Firmware and Configuration File Prefix and Postfix

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

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

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set "**Yes, every**" the auto check will be done in the minute specified in this field. If set to "**daily at hour (0-23)"**, Service Provider can use P193 (Auto Check Interval) to have the devices do a daily check at the hour set in this field with either Firmware Server or Config Server. If set to "**weekly on day (0-6)"** the auto check will be done in the day specified in this field. 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.

Automatic Upgrade:

○ No   ○ Yes, every [10080] minutes (60-5256000).

○ Yes, daily at hour [1] (0-23).   ○ Yes, weekly on day [1] (0-6).

RESTORE FACTORY DEFAULT SETTINGS

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

Using the Reset Button

To reset default factory settings using the reset button please follow the steps above:

1. Unplug the Ethernet cable.
2. Locate the reset hole on the back panel of your HT70X .
3. Insert a pin in this hole, and press for about 7 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

Using the IVR Command

Reset default factory settings using the IVR prompt:

1. Dial "***" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

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

Table 12: MAC Address Key Mapping

Key	Mapping
0-9	0-9
A	22 (press the "2" key twice, "A" will show on the LCD)
B	222
C	2222
D	33 (press the "3" key twice, "D" will show on the LCD)

	333	
	3333	

For example: if the MAC address is 000b8200e395, it should be keyed in as "0002228200333395".

EXPERIENCING HT70X

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 analogue telephone adapter, it will be sure to bring convenience to both your business and personal life.