HandyTone-701 User's Manual

Ver2.0.1 2016/02/08 created
Ver2.0.2 2016/08/23 revised

Thank you for purchasing Grandstream HT-701. Before using the product, please read the manual carefully. After reading, store the manual in an accessible place.

Check if all the accessories/attachments are included in the package.

- HT-701
- AC Adaptor
- Ethernet cable
- User's manual

DISCLAIMER

- Features and specifications may be updated without prior notice. Please bear with us.
- The company will not be responsible for the development of noise and volume interruption and disconnection of calls that arise from the customer and this company's network system. Please be advised that call quality is not guaranteed.
- The company will not be responsible for inconsistency in call quality brought about by individual listening skills.
- The company will not be responsible for the device damage that occurred outside Japan.
- Please acknowledge that services in HandyTone-701 may stop temporarily without prior notice due to HT- 701's scheduled maintenance.
- Please acknowledge that the company will not be responsible in the event that the phone can no longer be used for calling due to damage or malfunction.

ATTENTION

- Wipe the device with soft and dry cloth. Do not use wet towel.
- Do not use products with chemicals as these may cause damage to the device.
- If chemical dust cloth is used, read the accompanying directions carefully.

▪ HandyTone-701

HandyTone-701 is an adaptor used to connect an analog phone device to Agile Phone. To use the device, the customer needs to subscribe to an Internet Service Provider. Customer with no broadband router features needs to provide a separate broadband router.

Port	Description
INTERNET	Connects LAN cable's (Ethernet) accessory.
DC 12V	Connects special AC adaptor accessory.
PHONE	Connects analog phone device. ※ Use appropriate phone cable accessory to analog phone when connecting.
RESET	Use when restarting the device.

▪Connection Method

1. Switch the analog phone device's line type to tone.
2. Connect the phone cable with HandyTone-701's Phone and analog phone device.
3. Connect Ethernet cable with HandyTone-701's Internet to a broadband router.
4. Connect AC Adaptor to HandyTone-701's DC 12VI and plug to an electric socket.

※ Perform the procedure in correct order. Make sure that cable and adaptor were plugged in properly.

It takes a couple of minutes to place a call after connection. Please wait a moment.

① **Confirm HandyTone-701 device's IP address.**

1. After picking-up the handset of the phone device that is connected to HandyTone-701, dial ***.
2. If setting is not yet performed, E103E, an English voice, will be played from the handset. After dialing #, confirm whether an English voice saying ENTER A MENU OPTION is played.
   ※ If already set, a voice saying ENTER A MENU OPTION will be played.
3. After ENTER A MENU OPTION is played, dial 02. An English voice telling HandyTone-701's IP address will be played.
   Example: 1-9-2.1-6-8.1.2

② **From web browser, open Setting page.**

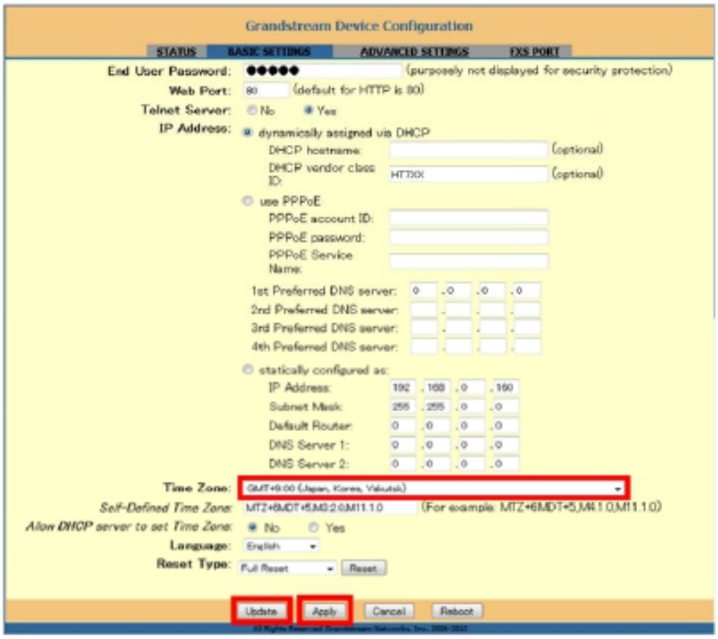
1. From the HandyTone-701 and computers connected within the same network, when launching Internet Explorer, etc. web browser, IP address should be entered following the format below.

http://(IP Address)/

Example: If 192.168.1.2 is displayed, it should be entered as http://192.168.1.2/

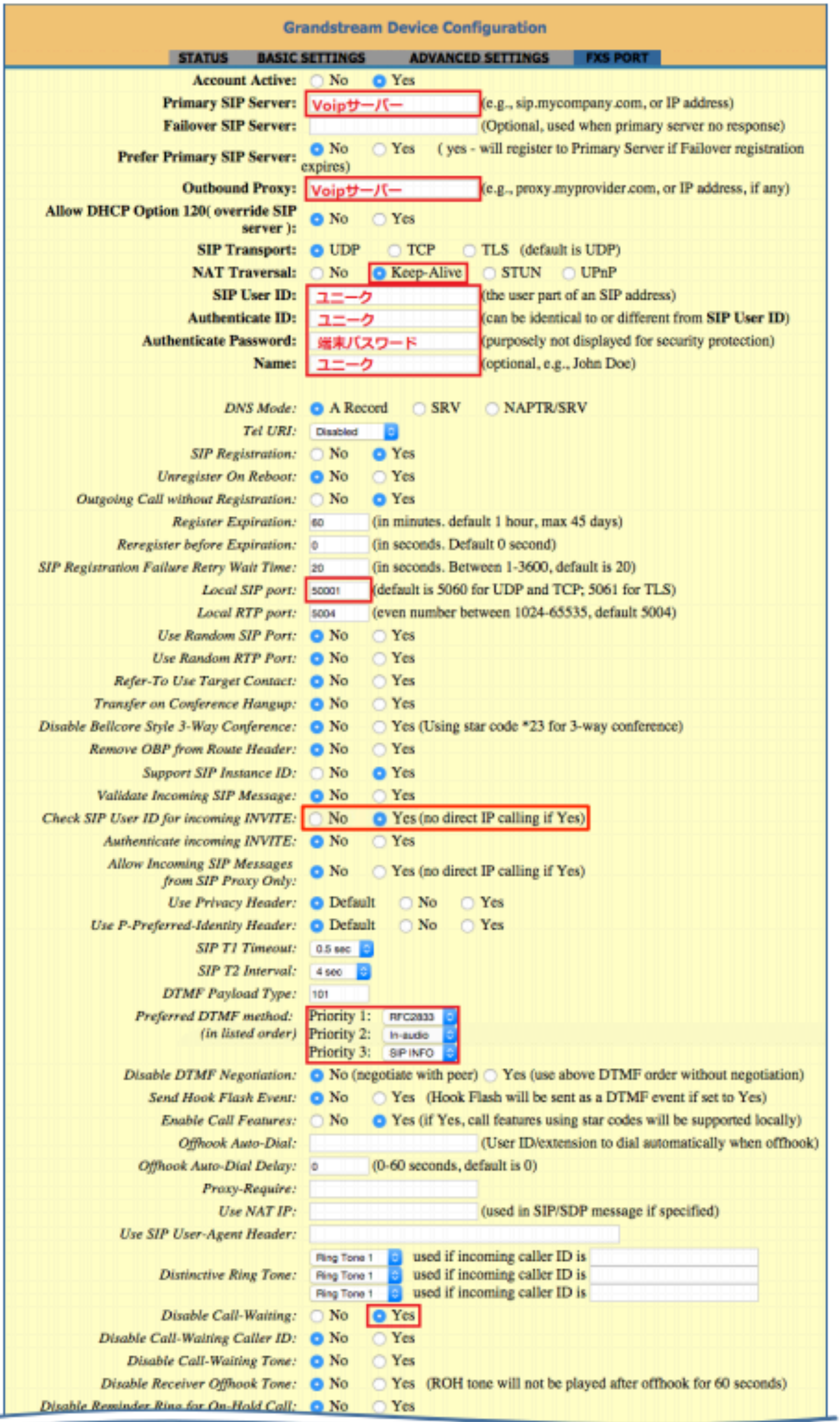
2. The above page will be displayed. Enter password and click Login. The initial password is admin.

③ **Click the BASIC SETTINGS tab found at the upper portion of the page.**



Time Zone： Select GMT+9:00 (Japan,Korea,Yakutsk).
Once done, click Update and Apply respectively.

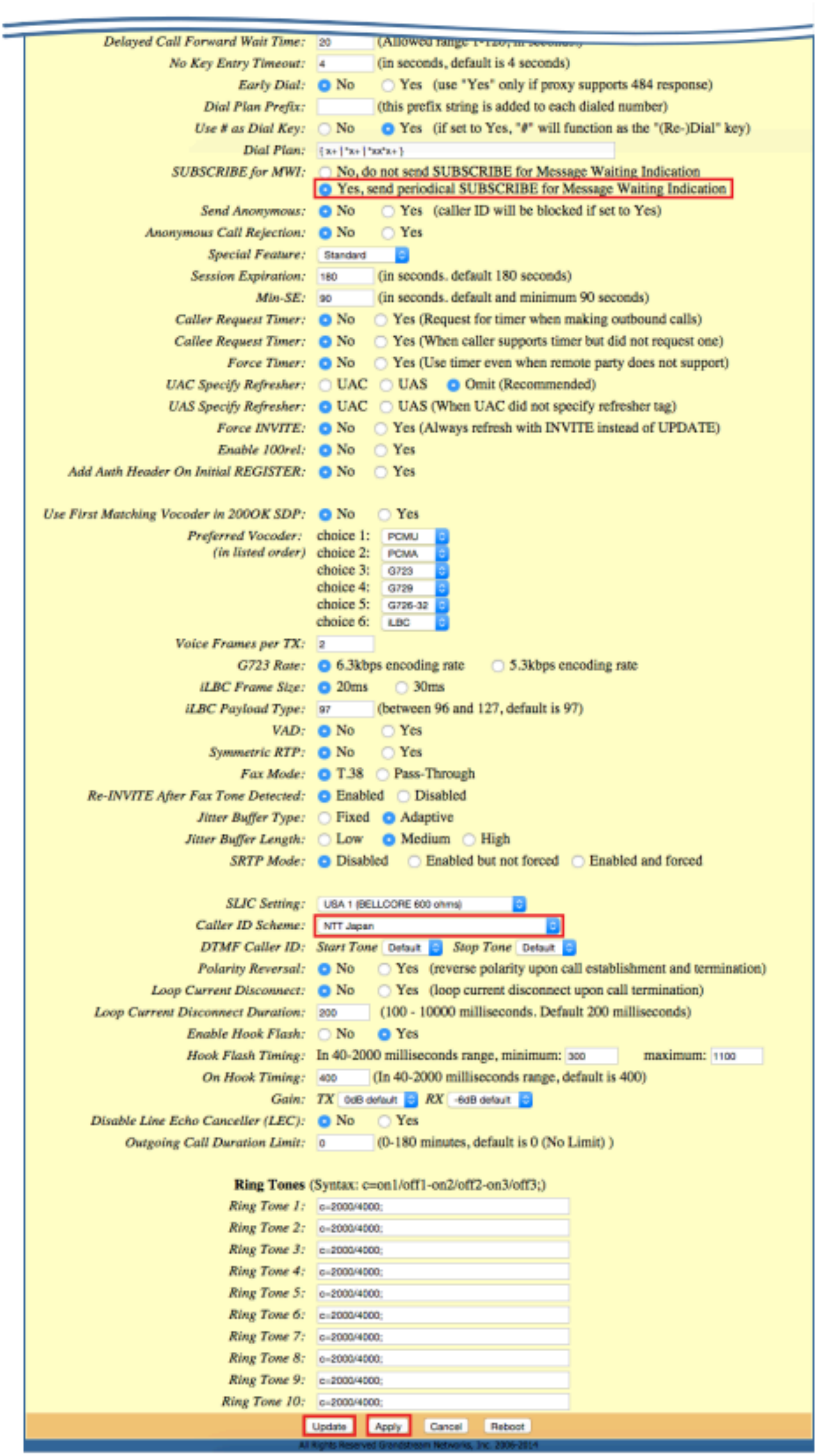
④ Click the FXS PORT tab found at the upper portion of the page.



Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | FXS PORT

Account Active:	○ No ● Yes
Primary SIP Server:	Voipサーバー (e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	(Optional, used when primary server no response)
Prefer Primary SIP Server:	● No ○ Yes (yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	Voipサーバー (e.g., proxy.myprovider.com, or IP address, if any)
Allow DHCP Option 120(override SIP server):	● No ○ Yes
SIP Transport:	● UDP ○ TCP ○ TLS (default is UDP)
NAT Traversal:	○ No ● Keep-Alive ○ STUN ○ UPnP
SIP User ID:	ユニーク (the user part of an SIP address)
Authenticate ID:	ユニーク (can be identical to or different from **SIP User ID**)
Authenticate Password:	端末パスワード (purposely not displayed for security protection)
Name:	ユニーク (optional, e.g., John Doe)
DNS Mode:	● A Record ○ SRV ○ NAPTR/SRV
Tel URI:	Disabled
SIP Registration:	○ No ● Yes
Unregister On Reboot:	● No ○ Yes
Outgoing Call without Registration:	○ No ● Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
Reregister before Expiration:	0 (in seconds. Default 0 second)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
Local SIP port:	50001 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP port:	5004 (even number between 1024-65535, default 5004)
Use Random SIP Port:	● No ○ Yes
Use Random RTP Port:	● No ○ Yes
Refer-To Use Target Contact:	● No ○ Yes
Transfer on Conference Hangup:	● No ○ Yes
Disable Bellcore Style 3-Way Conference:	● No ○ Yes (Using star code *23 for 3-way conference)
Remove OBP from Route Header:	● No ○ Yes
Support SIP Instance ID:	○ No ● Yes
Validate Incoming SIP Message:	● No ○ Yes
Check SIP User ID for incoming INVITE:	○ No ● Yes (no direct IP calling if Yes)
Authenticate incoming INVITE:	● No ○ Yes
Allow Incoming SIP Messages from SIP Proxy Only:	● No ○ Yes (no direct IP calling if Yes)
Use Privacy Header:	● Default ○ No ○ Yes
Use P-Preferred-Identity Header:	● Default ○ No ○ Yes
SIP T1 Timeout:	0.5 sec
SIP T2 Interval:	4 sec
DTMF Payload Type:	101
Preferred DTMF method: (in listed order)	Priority 1: RFC2833 Priority 2: In-audio Priority 3: SIP INFO
Disable DTMF Negotiation:	● No (negotiate with peer) ○ Yes (use above DTMF order without negotiation)
Send Hook Flash Event:	● No ○ Yes (Hook Flash will be sent as a DTMF event if set to Yes)
Enable Call Features:	○ No ● Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Offhook Auto-Dial Delay:	0 (0-60 seconds, default is 0)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
Use SIP User-Agent Header:	
Distinctive Ring Tone:	Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is
Disable Call-Waiting:	○ No ● Yes
Disable Call-Waiting Caller ID:	● No ○ Yes
Disable Call-Waiting Tone:	● No ○ Yes
Disable Receiver Offhook Tone:	● No ○ Yes (ROH tone will not be played after offhook for 60 seconds)
Disable Reminder Ring for On-Hold Call:	● No ○ Yes



Delayed Call Forward Wait Time: 20 (Allowed range 1-120, in seconds)
No Key Entry Timeout: 4 (in seconds, default is 4 seconds)
Early Dial: No / Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix: (this prefix string is added to each dialed number)
Use # as Dial Key: No / Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan: { x+ | *x+ | *xx*x+ }
SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication / Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous: No / Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection: No / Yes
Special Feature: Standard
Session Expiration: 180 (in seconds. default 180 seconds)
Min-SE: 90 (in seconds. default and minimum 90 seconds)
Caller Request Timer: No / Yes (Request for timer when making outbound calls)
Callee Request Timer: No / Yes (When caller supports timer but did not request one)
Force Timer: No / Yes (Use timer even when remote party does not support)
UAC Specify Refresher: UAC / UAS / Omit (Recommended)
UAS Specify Refresher: UAC / UAS (When UAC did not specify refresher tag)
Force INVITE: No / Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel: No / Yes
Add Auth Header On Initial REGISTER: No / Yes

Use First Matching Vocoder in 200OK SDP: No / Yes
Preferred Vocoder: (in listed order) choice 1: PCMU; choice 2: PCMA; choice 3: G723; choice 4: G729; choice 5: G726-32; choice 6: iLBC
Voice Frames per TX: 2
G723 Rate: 6.3kbps encoding rate / 5.3kbps encoding rate
iLBC Frame Size: 20ms / 30ms
iLBC Payload Type: 97 (between 96 and 127, default is 97)
VAD: No / Yes
Symmetric RTP: No / Yes
Fax Mode: T.38 / Pass-Through
Re-INVITE After Fax Tone Detected: Enabled / Disabled
Jitter Buffer Type: Fixed / Adaptive
Jitter Buffer Length: Low / Medium / High
SRTP Mode: Disabled / Enabled but not forced / Enabled and forced

SLIC Setting: USA 1 (BELLCORE 600 ohms)
Caller ID Scheme: NTT Japan
DTMF Caller ID: Start Tone Default Stop Tone Default
Polarity Reversal: No / Yes (reverse polarity upon call establishment and termination)
Loop Current Disconnect: No / Yes (loop current disconnect upon call termination)
Loop Current Disconnect Duration: 200 (100 - 10000 milliseconds. Default 200 milliseconds)
Enable Hook Flash: No / Yes
Hook Flash Timing: In 40-2000 milliseconds range, minimum: 300 maximum: 1100
On Hook Timing: 400 (In 40-2000 milliseconds range, default is 400)
Gain: TX 0dB default RX -6dB default
Disable Line Echo Canceller (LEC): No / Yes
Outgoing Call Duration Limit: 0 (0-180 minutes, default is 0 (No Limit))

Ring Tones (Syntax: c=on1/off1-on2/off2-on3/off3;)
Ring Tone 1: c=2000/4000;
Ring Tone 2: c=2000/4000;
Ring Tone 3: c=2000/4000;
Ring Tone 4: c=2000/4000;
Ring Tone 5: c=2000/4000;
Ring Tone 6: c=2000/4000;
Ring Tone 7: c=2000/4000;
Ring Tone 8: c=2000/4000;
Ring Tone 9: c=2000/4000;
Ring Tone 10: c=2000/4000;

Update Apply Cancel Reboot

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Primary SIP Server : Enter VoIP server.

Outbound Proxy : Enter VoIP server.

NAT Traversal : Select keep-alive.

SIP User ID： Enter Unique (10 numeric characters)

Authenticate ID： Enter Unique (10 numeric characters)

Authenticate Password： Enter device password.

Name： Enter Unique (10 numeric characters)

Local SIP Port: any Port number in range of 50000~60000
*Please allocate different port number to each devices without overlapping.
Ex.) Device A : 50000, Device B : 500001

Check SIP User ID for Incoming INVITE : [Yes]
*It will be blocked unauthorized access to the devices.

Preferred DTMF method： Supply Priority 1, 2 and 3 with the same information as below:
Priority 1： RFC2833
Priority 2： In-audio
Priority 3： SIP INFO
*Set [in-audio] as Priority 1 in case [DTMF inband] are set on Unique Management Page.

Disable Call-Waiting：
- Selecting Yes will notify new caller that line is busy when the callee is still at the middle of the call.
- Selecting No will notify new caller to assume that the line is ringing/active even when the callee is at the middle of a call.

※ This is an optional setting. No setting required if no need.
※ Default is [No]. Factory setting by the company : [Yes]

SUBSCRIBE for MWI : Select [Yes, send periodical SUBSCRIBE for Message Waiting Indication]

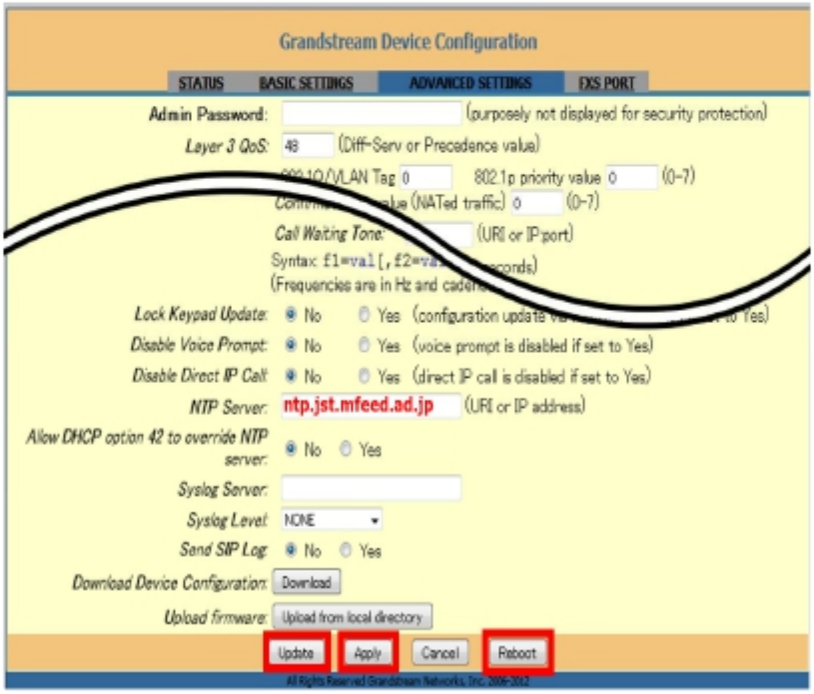
Caller ID Scheme： Set using the example below.
- NTT Japan： Select when connecting to a phone device that supports number display

※ It is important to set phone that supports number display. Please refer to the device's instruction manual for the setup method.
- Bellcore/Telcordia： Select this when using phone other than NTT Japan.

※ The prescribed value is Bellcore/Telcordia. The default factory setting is NTT Japan. If the device does not support number display, change the setting to Bellcore/Telcordia.

Once done, click Update and Apply respectively.

⑤ Click the ADVANCED SETTINGS tab found at the upper portion of the page.



NTP Server： Enter ntp.jst.mfeed.ad.jp.
Once done, click Update and Apply respectively. Click Reboot. The device will be restarted and settings will be applied.

・ Important points when using Agile Network
Always disable the alert info header setting in Unique that you are using on the device. Incoming call will be blocked when this setting is enabled.

From website, open Unique Management page.
Click Unique Setting, then, go to General Setting.
In Alert Info Header Setting, select DISABLE.

▪ Confirmation Setting from English Voice Guidance

1. Pick-up the handset.
2. From HandyTone-701 connected phone device, enter * * *. From the handset, a voice guidance saying ENTER A MENU OPTION will be played. When you dial *, confirming the status, DHCP mode or STATIC IP mode, is possible.
3. Enter menu number from phone device and perform confirmation of setting.
4. As needed, dial 9 and set the operation.

Menu Number	English Voice Guidance	Operating procedure/Contents
01	DHCP MODE (Dynamic Host Configuration Protocol Mode) and STATIC IP MODE	Warning: Static IP mode is not supported in Agile Phone service.
02	IP ADDRESS	Voice guidance for the current HandyTone's IP address will be played.
03	SUBNET	The current subnet mask's IP address will be played in voice guidance.
04	GATEWAY	Voice guidance for the current gateway's IP address will be played.
05	DNS SERVER	Voice guidance for the current DNS server's IP address will be played.
	INVALID ENTRY	Returns to Main Menu automatically.

This manual contains precautions to assure user's safety while using this product.
If the precautions are disregarded, the extent of consequences is presented below.

WARNING — This indicates that ignoring or mishandling of this notice might result to death or serious injury to the person.

CAUTION — This indicates that ignoring or mishandling of this notice might result to harm to the person.

WARNING

- Immediately unplug the device from the power supply if there is a strange smell and smoke. Don't use the device and immediately contact the dealer for repair. This may cause fire or explosion.
- Do not disassemble or modify. The device may malfunction, cause electric shock and fire.
- Do not insert metal to the opening or drop the product. If in case a metal is inserted to the device, unplug the device and contact the dealer. The device may malfunction, cause electric shock and fire.
- Do not pour water or any foreign object inside the device. If in case water or any foreign object is poured or inserted to the device, unplug the device and contact the dealer.
- Do not touch the device when lightning occurs. This may cause electric shock.
- Do not use in bathroom or shower room. The device may malfunction, cause electric shock and fire.
- Don't place in a damp, dusty or direct sunlight. The device may malfunction, cause electric shock and fire.
- This device is not waterproof. If in case water entered the device, unplug the device and contact the dealer.

CAUTION

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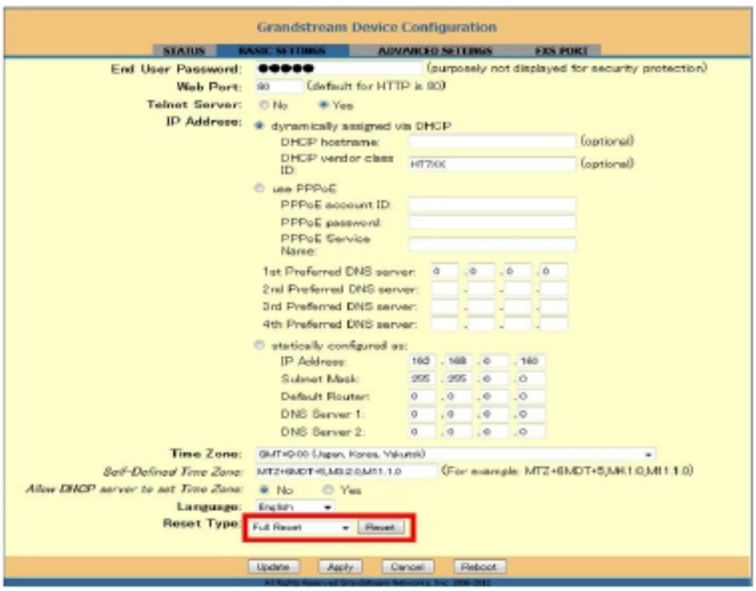
PROBLEM	CAUSE	SOLUTION
General Failure	Please plug and unplug from the power source. (Do this a few times). Please wait at least one minute before you again plug the device to the power source.	
No ringer sound	Ethernet cable is disconnected. AC adapter is disconnected. Did you change the volume settings? Did you change the settings on the display”	Please check the connection of the Ethernet cable. Please check the connection of the AC adapter. Please adjust the volume on the button. Revert the change settings. Disconect the Ethernet cable (reboot).
Call is interrupted by noise	Are there ani abnormalities in the network environment?	Please check your network environment.

1. From web browser, open Setting page and Login.
2. Click BASIC SETTINGS tab found at the upper portion of the page.



3. Reset Type: Immediately after selecting Full Reset, press the Reset button.
4. The phone device will be restarted and the phone will be initializing.