

# Grandstream Gateways & ATA's

HT-286, 502, 503, 701, 702, 704, GWX-4008 & 4004

## Admin Guide

- MANUAL CONFIGURATION





## Find the IP Address

### Quick Steps

1. Lift the handset of the phone connected to your ATA, Dial \*\*\* to initiate the IVR Menu
2. Wait to be instructed to enter your command.
3. Dial 02 and the IP address will be played to you.

## MANUAL CONFIGURATION

### Quick Steps

1. Find the IP address of the device and open a web browser to the IP address.
2. Login with the device admin password. Default factory password is admin

A screenshot of a web browser displaying the login page for a Grandstream device. The page has a blue header with the text "Grandstream Device Configuration". Below the header is a yellow section containing a "Password" label and a text input field. Below the input field is a "Login" button. At the bottom of the page, there is a small copyright notice: "All Rights Reserved Grandstream Networks, Inc. 2006-2011".

Grandstream Device Configuration

Password

Login

All Rights Reserved Grandstream Networks, Inc. 2006-2011

- Click on the “**Basic Settings**” option at the top.

Your telephones receive their time from the ATA device. Be sure you select the correct time zone below and enter NTP server value (in Advanced Settings):

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT

**End User Password:**  (purposely not displayed for security protection)

**Web Port:**  (default for HTTP is 80)

**Telnet Server:**  No  Yes

**IP Address:**  dynamically assigned via DHCP

DHCP hostname:  (optional)

DHCP vendor class ID:  (optional)

use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

1st Preferred DNS server:

2nd Preferred DNS server:

3rd Preferred DNS server:

4th Preferred DNS server:

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

**Time Zone:** GMT-05:00 (Eastern Time) ←

*Self-Defined Time Zone:*  (For example: MTZ+6MDT+5,M4.1.0,M11.1.0)

*Allow DHCP server to set Time Zone:*  No  Yes ←

**Language:**

**Reset Type:**

Click Apply.



4. Click on the “**Advanced Settings**” option at the top.

To prevent your ATA device from receiving direct IP calls (calls not originating through VoIP Much), set the following option to Yes:

Disable Direct IP Call = Yes

Your telephones receive their time from the ATA device. Be sure you set the correct time zone (in Basic Settings) and the below NTP server value:

NTP Server: ca.pool.ntp.org

Lock Keypad Update:  No  Yes (configuration update via keypad is disabled if set to Yes)  
Disable Voice Prompt:  No  Yes (voice prompt is disabled if set to Yes)  
Disable Direct IP Call:  No  Yes (direct IP call is disabled if set to Yes)  
NTP Server:  (URI or IP address)  
Allow DHCP option 42 to override NTP server:  No  Yes  
Syslog Server:   
Syslog Level: NONE ▾  
Send SIP Log:  No  Yes  
Download Device Configuration:   
Upload firmware:

Click Apply



5. Click on the “**FXS Port**” option at the top.

Primary SIP Server = SIP Server Address from your SETUP PDF

SIP User ID = ATA Device Username from your SETUP PDF

Authenticate ID = ATA Device Username from your SETUP PDF

Authenticate Password = ATA Device Password from your SETUP PDF

**Grandstream Device Configuration**

STATUSBASIC SETTINGSADVANCED SETTINGSFXS PORT

**Account Active:**  No  Yes

**Primary SIP Server:**  (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:**  (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:**  (the user part of an SIP address)

**Authenticate ID:**  (can be identical to or different from SIP User ID)

**Authenticate Password:**  (purposely not displayed for security protection)

**Name:**  (optional, e.g., John Doe)

**DNS Mode:**  A Record  SRV  NAPTR/SRV

**Tel URI:**

**SIP Registration:**  No  Yes

**Unregister On Reboot:**  No  Yes

**Outgoing Call without Registration:**  No  Yes


**Register Expiration:**  (in minutes. default 1 hour, max 45 days)

**Reregister before Expiration:**  (in seconds. Default 0 second)



To prevent your ATA device from receiving direct IP calls (calls not originating through VoIP Much), set the following option to Yes:

Allow Incoming SIP Messages from SIP Proxy Only = Yes

<i>SIP Registration Failure Retry Wait Time:</i>	<input type="text" value="20"/>	(in seconds. Between 1-3600, default is 20)
<i>Layer 3 QoS:</i>	<input type="text" value="26"/>	SIP DSCP (Diff-Serv value in decimal, default 24)
	<input type="text" value="46"/>	RTP DSCP (Diff-Serv value in decimal, default 46)
<i>Local SIP port:</i>	<input type="text" value="5060"/>	(default is 5060 for UDP and TCP; 5061 for TLS)
<i>Local RTP port:</i>	<input type="text" value="5004"/>	(even number between 1024-65535, default 5004)
<i>Use Random SIP Port:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Use Random RTP Port:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Refer-To Use Target Contact:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Transfer on Conference Hangup:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Disable Bellcore Style 3-Way Conference:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes (Using star code *23 for 3-way conference)
<i>Remove OBP from Route Header:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Support SIP Instance ID:</i>	<input type="radio"/> No	<input checked="" type="radio"/> Yes
<i>Validate Incoming SIP Message:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Check SIP User ID for incoming INVITE:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes (no direct IP calling if Yes)
<i>Authenticate incoming INVITE:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes
<i>Allow Incoming SIP Messages from SIP Proxy Only:</i>	<input type="radio"/> No	<input checked="" type="radio"/> Yes (no direct IP calling if Yes) 
<i>Use Privacy Header:</i>	<input checked="" type="radio"/> Default	<input type="radio"/> No <input type="radio"/> Yes
<i>Use P-Preferred-Identity Header:</i>	<input checked="" type="radio"/> Default	<input type="radio"/> No <input type="radio"/> Yes
<i>SIP REGISTER Contact Header Uses:</i>	<input checked="" type="radio"/> LAN Address	<input type="radio"/> WAN Address
<i>SIP T1 Timeout:</i>	<input type="text" value="0.5 sec"/>	
<i>SIP T2 Interval:</i>	<input type="text" value="4 sec"/>	
<i>SIP Timer D:</i>	<input type="text" value="0"/>	(0 - 64 seconds. Default 0)
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>	
<i>Preferred DTMF method:</i>	Priority 1:	<input type="text" value="RFC2833"/>
<i>(in listed order)</i>	Priority 2:	<input type="text" value="SIP INFO"/>
	Priority 3:	<input type="text" value="In-audio"/>
<i>Disable DTMF Negotiation:</i>	<input checked="" type="radio"/> No (negotiate with peer)	<input type="radio"/> Yes (use above DTMF order without negotiation)

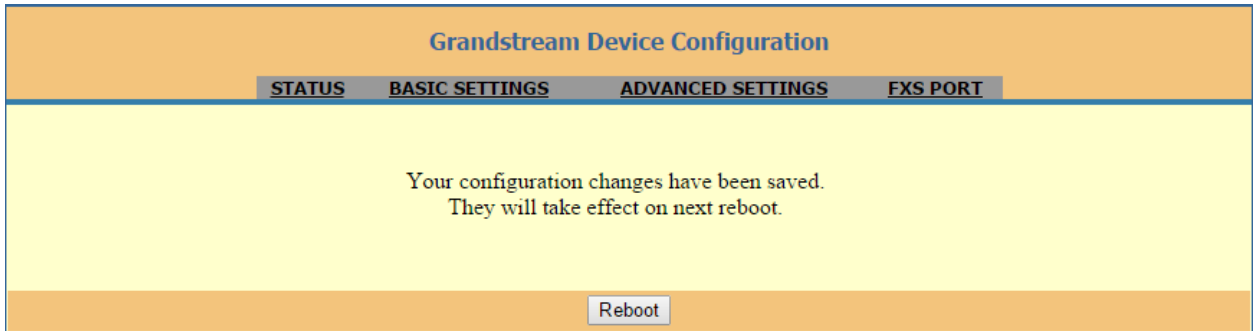
The remaining settings on this page remain at their default values.

Click Apply.

<input type="button" value="Update"/>	<input type="button" value="Apply"/>	<input type="button" value="Cancel"/>	<input type="button" value="Reboot"/>
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6. Click Reboot



Your ATA will reboot and register to the VoIP Much network. Reboot time can be approx. 1 to 2 minutes.