Grandstream Gateways & ATA’s

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

Admin Guide

- **MANUAL CONFIGURATION**

Find the IP Address

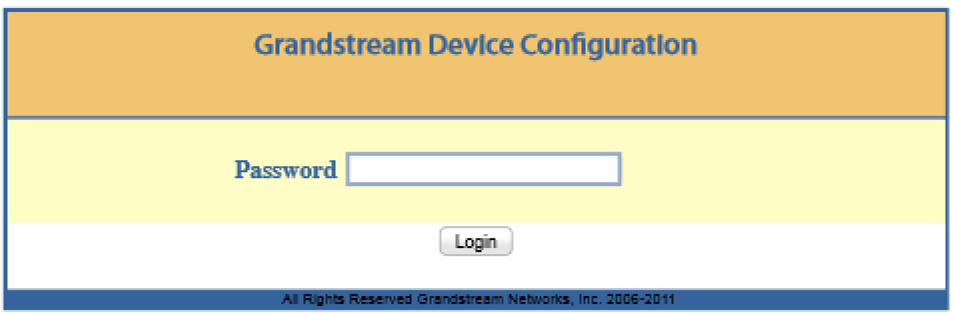
Quick Steps

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

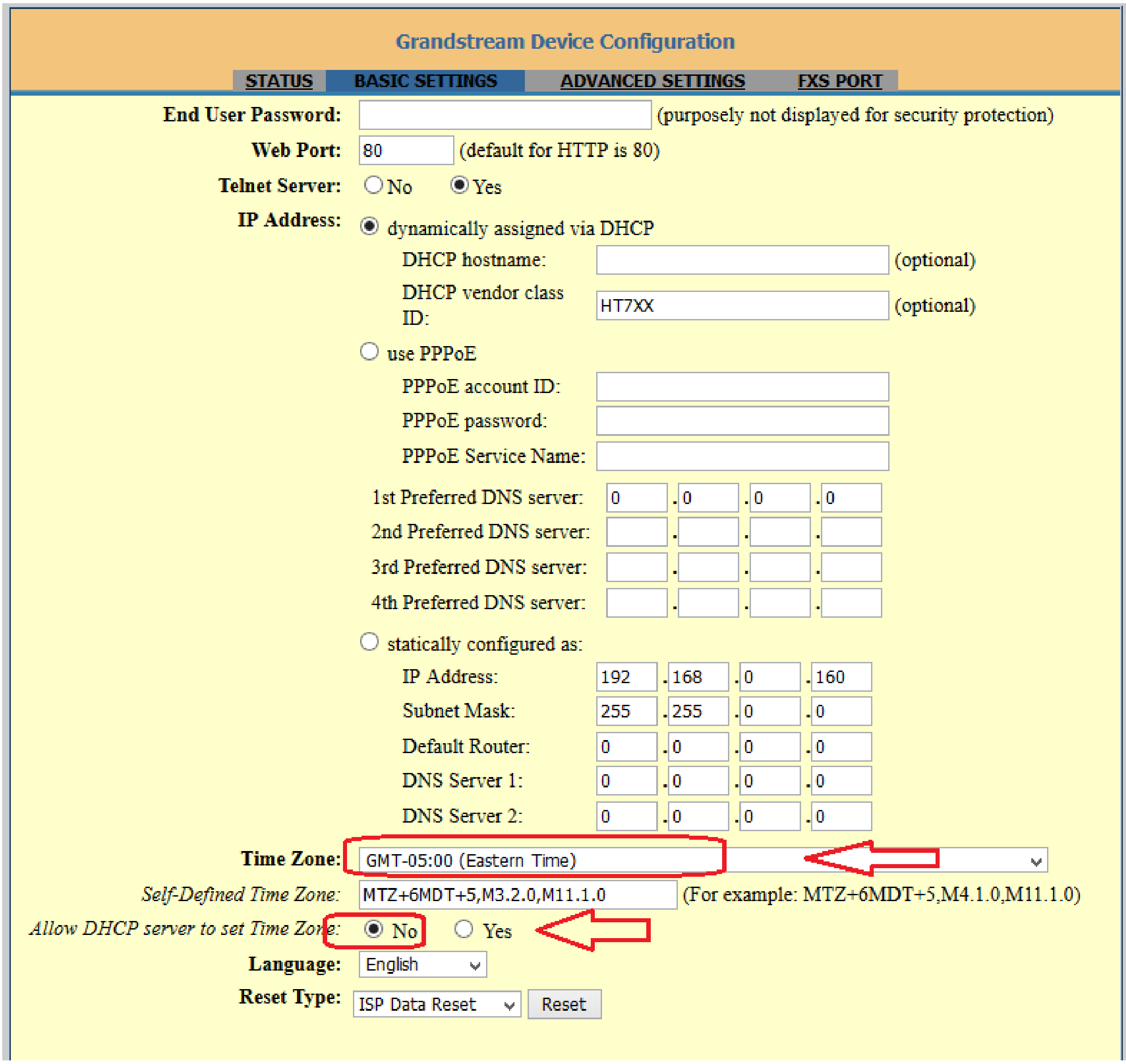
Quick Steps

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

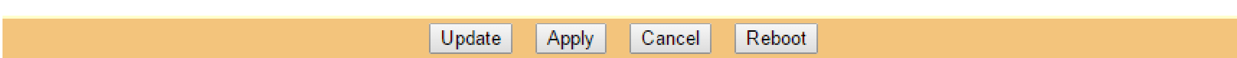


3. Click on the "**Basic Settings**" option at the top.

   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):



Click Apply.



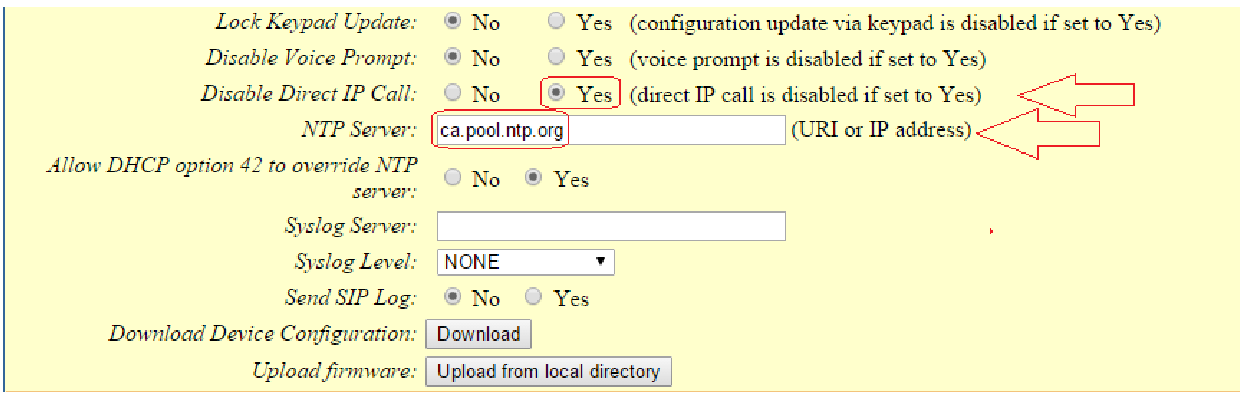
4. Click on the “**Advanced Settings**” option at the top.

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:

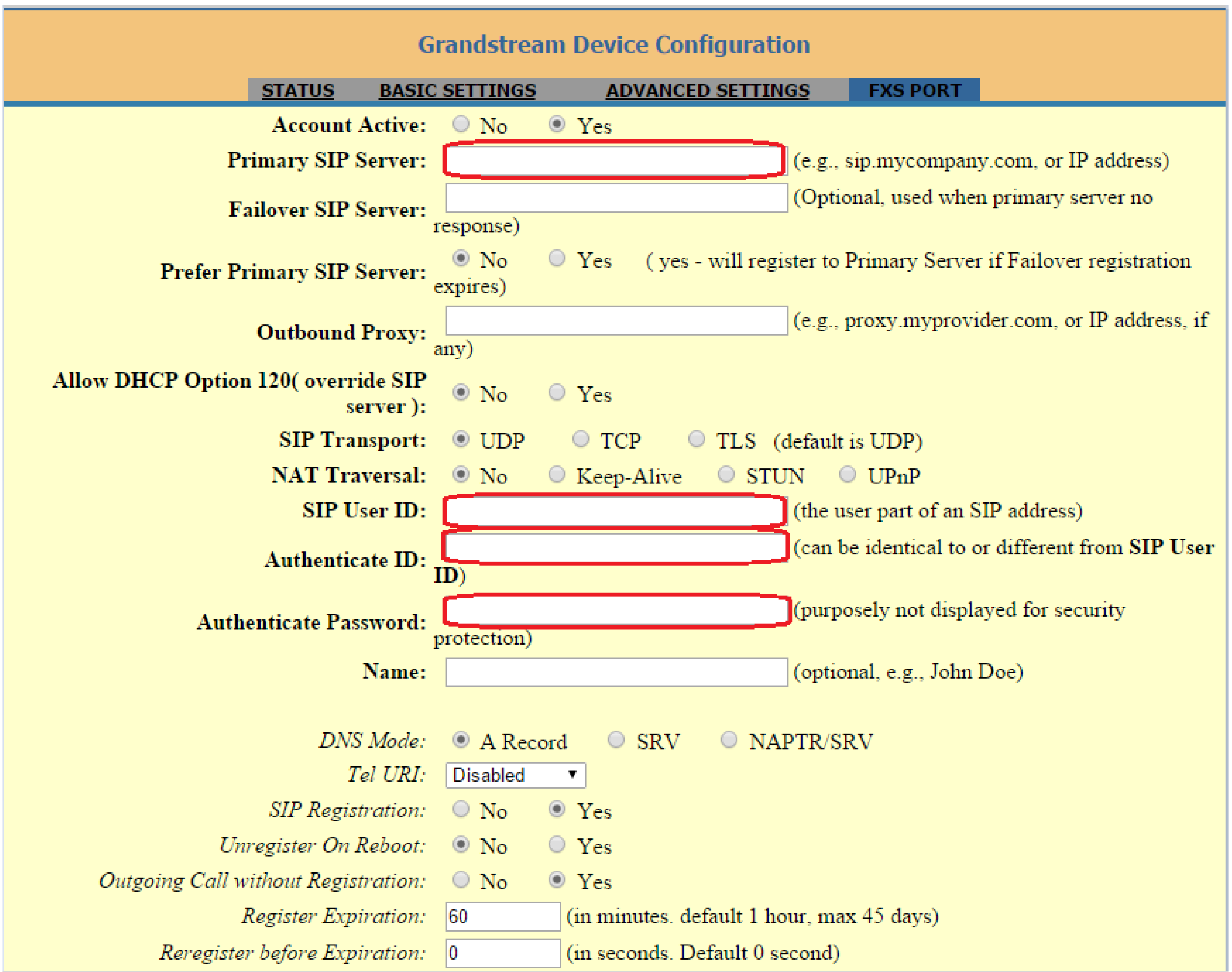
NTP Server: ca.pool.ntp.org



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



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

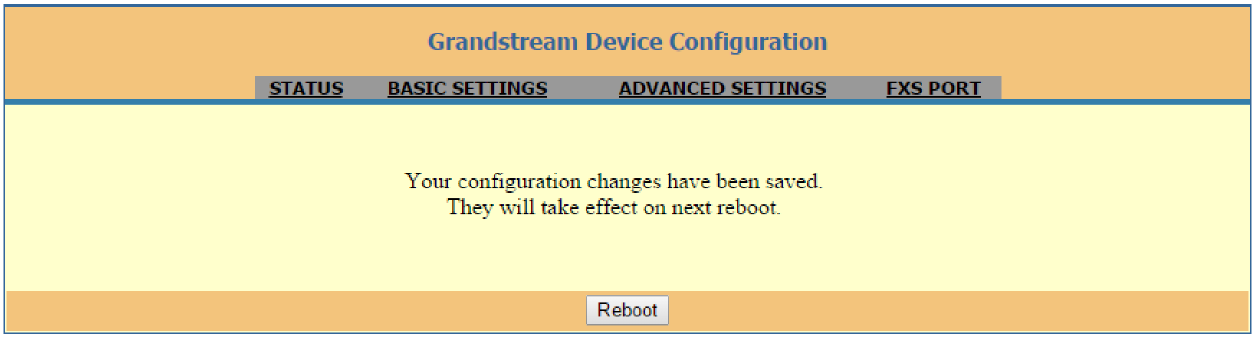
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, default 24) 46 RTP DSCP (Diff-Serv value in decimal, default 46)
Local SIP port:	5060 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP port:	5004 (even 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 REGISTER Contact Header Uses:	◉ LAN Address ○ WAN Address
SIP T1 Timeout:	0.5 sec ▼
SIP T2 Interval:	4 sec ▼
SIP Timer D:	0 (0 - 64 seconds. Default 0)
DTMF Payload Type:	101
Preferred DTMF method: (in listed order)	Priority 1: RFC2833 ▼ Priority 2: SIP INFO ▼ Priority 3: In-audio ▼
Disable DTMF Negotiation:	◉ No (negotiate with peer) ○ Yes (use above DTMF order without negotiation)

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

Click Apply.

Update Apply Cancel Reboot

6. Click Reboot



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