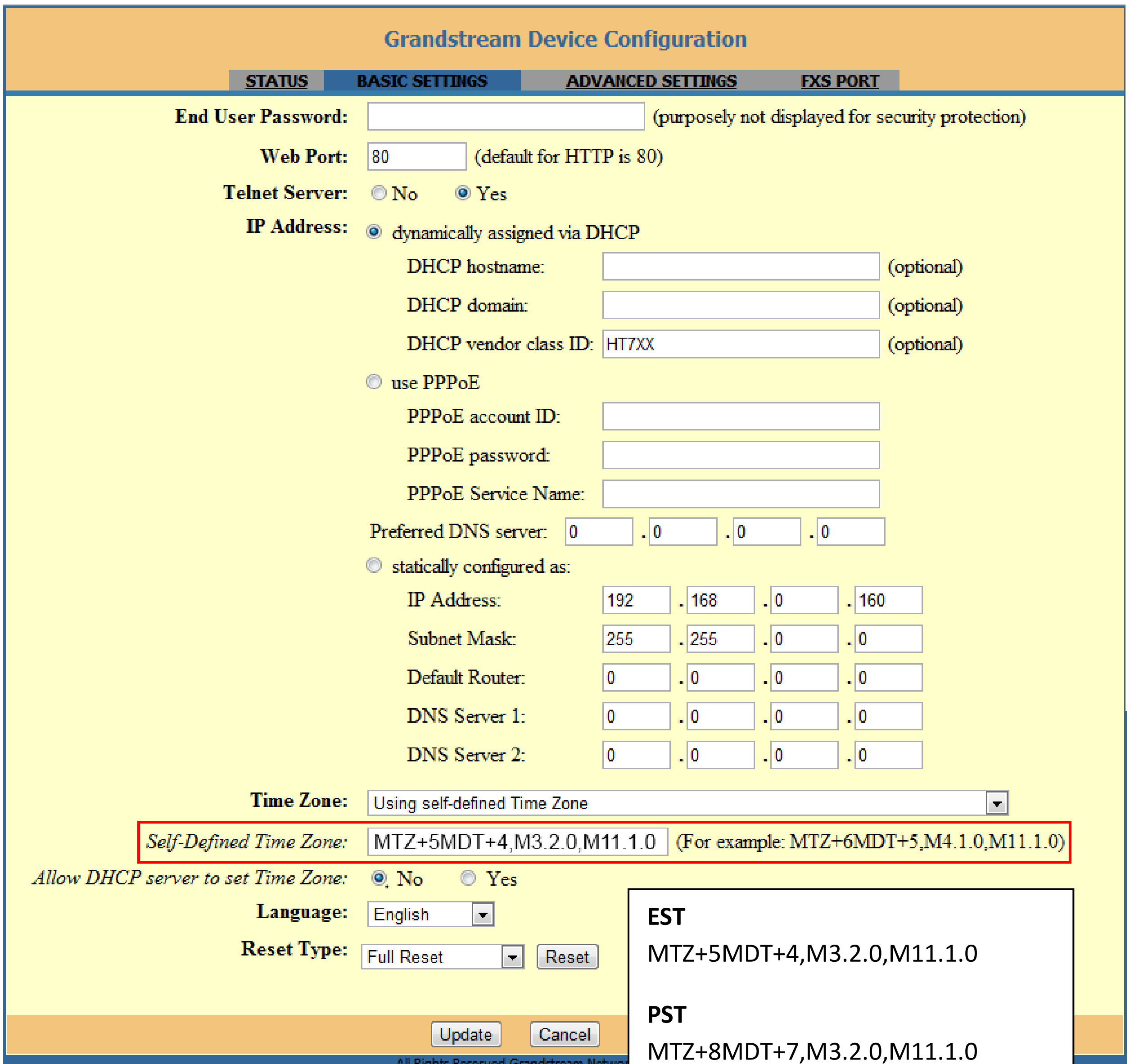
Grandstream HT-701 Configuration

Firmware 1.0.0.18 and lower

To access the web interface: dial '***' then '**02**' from the phone attached to the ATA.
Next, enter the IP address provided in your web browser. Login password is **admin**

On the 'Basic Settings' page modify the 'Self-Defined Time Zone' box to set the appropriate time zone for your location. Time zone strings are provided below the screenshot.



EST
MTZ+5MDT+4,M3.2.0,M11.1.0

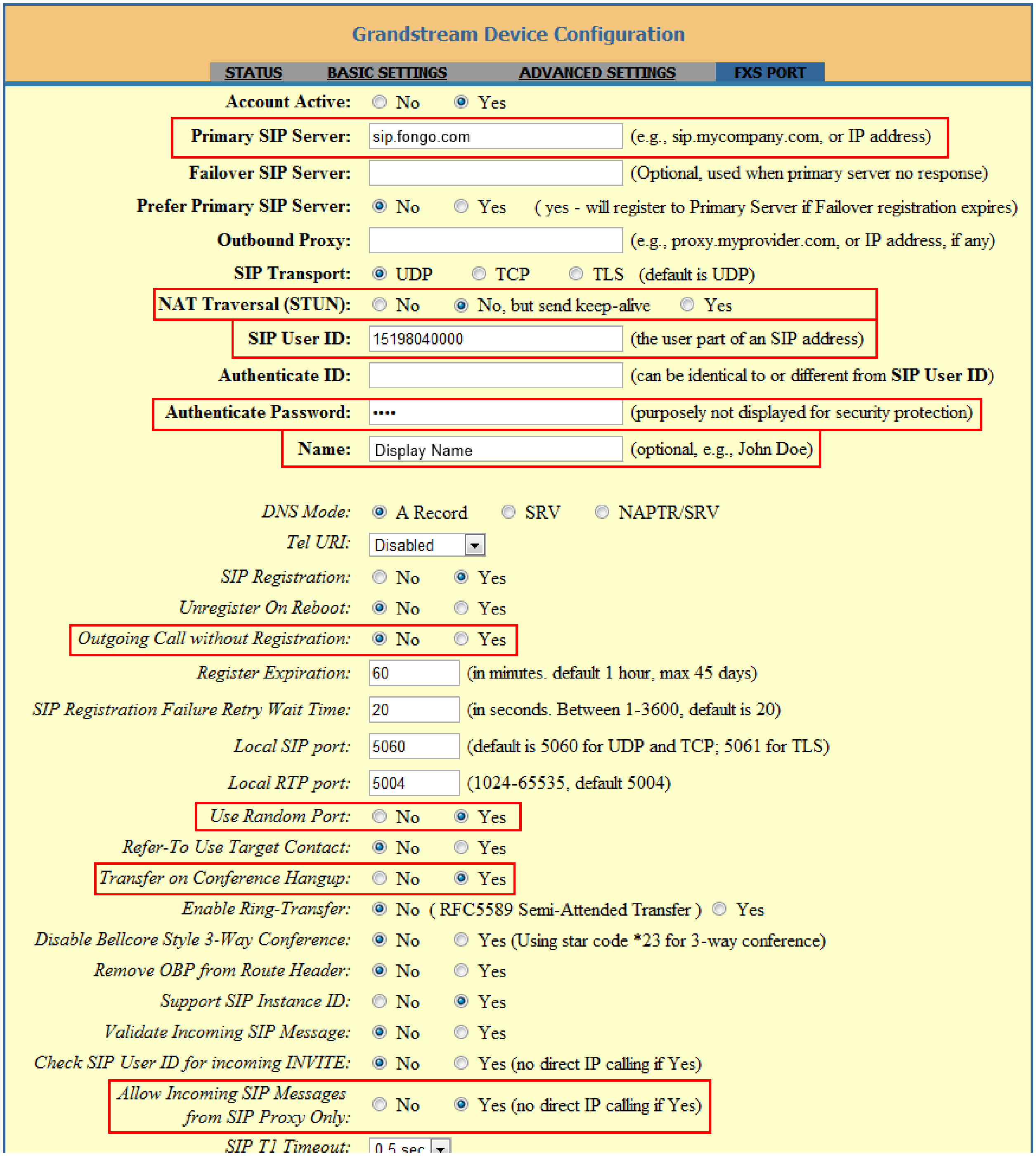
PST
MTZ+8MDT+7,M3.2.0,M11.1.0

MST
MTZ+7MDT+6,M3.2.0,M11.1.0

CST
MTZ+6MDT+5,M3.2.0,M11.1.0

Next, access the FXS Port page and update the highlighted fields:

Note: SIP User ID, Authenticate Password and Name will be unique for each user.



Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | FXS PORT

Account Active: No / Yes

Primary SIP Server: sip.fongo.com (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)

SIP Transport: UDP / TCP / TLS (default is UDP)

NAT Traversal (STUN): No / No, but send keep-alive / Yes

SIP User ID: 15198040000 (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: Display Name (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)

SIP Registration Failure Retry Wait Time: 20 (in seconds. Between 1-3600, default is 20)

Local SIP port: 5060 (default is 5060 for UDP and TCP; 5061 for TLS)

Local RTP port: 5004 (1024-65535, default 5004)

Use Random Port: No / Yes

Refer-To Use Target Contact: No / Yes

Transfer on Conference Hangup: No / Yes

Enable Ring-Transfer: No (RFC5589 Semi-Attended Transfer) / Yes

Disable Bellcore Style 3-Way Conference: No / Yes (Using star code *23 for 3-way conference)

Remove OBP from Route Header: No / Yes

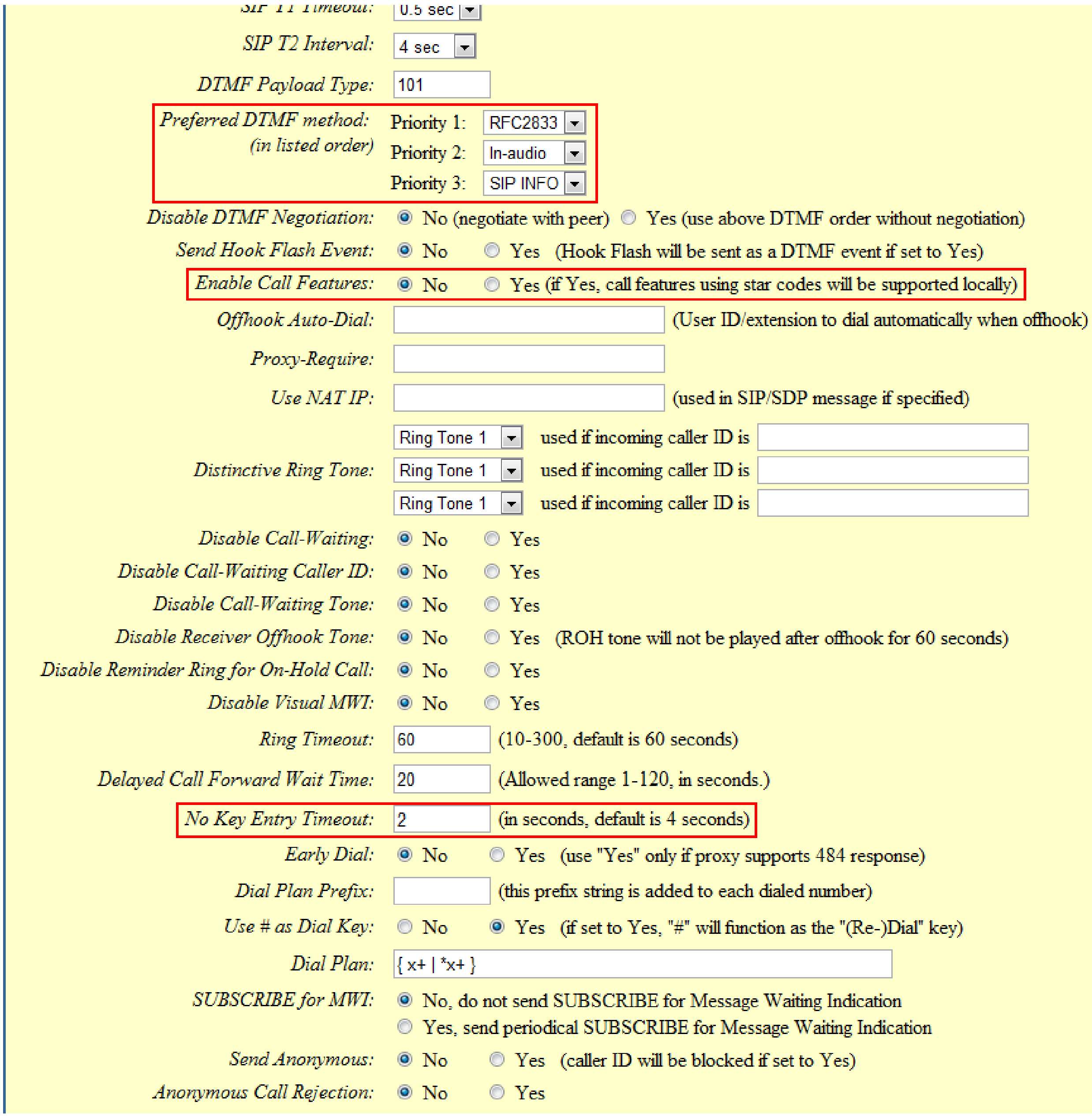
Support SIP Instance ID: No / Yes

Validate Incoming SIP Message: No / Yes

Check SIP User ID for incoming INVITE: No / Yes (no direct IP calling if Yes)

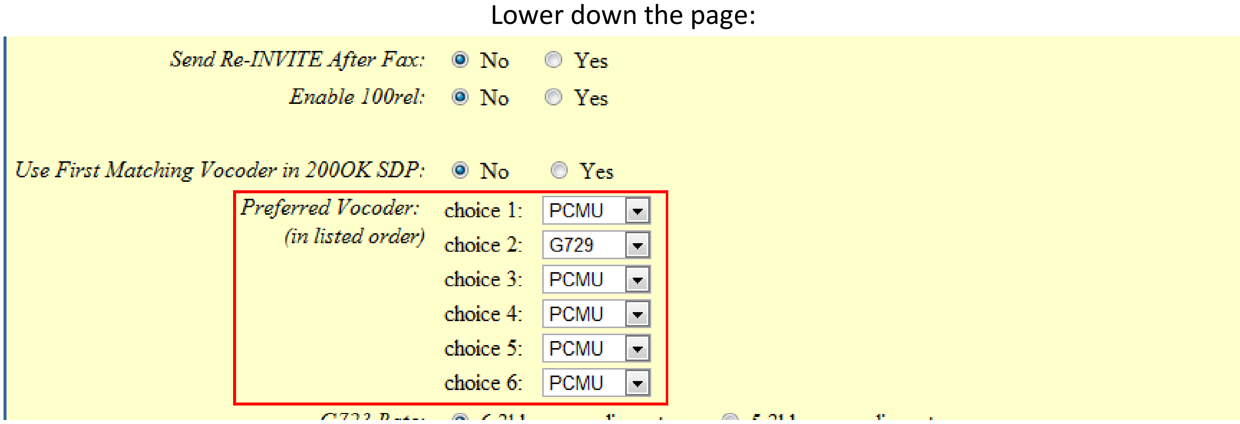
Allow Incoming SIP Messages from SIP Proxy Only: No / Yes (no direct IP calling if Yes)

SIP T1 Timeout: 0.5 sec



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)
Proxy-Require:
Use NAT IP: (used in SIP/SDP message if specified)
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
Disable Visual MWI: ◉ No ○ Yes
Ring Timeout: 60 (10-300, default is 60 seconds)
Delayed Call Forward Wait Time: 20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout: 2 (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+ }
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

Lower down the page:



Send Re-INVITE After Fax: ◉ No ○ Yes
Enable 100rel: ◉ No ○ Yes
Use First Matching Vocoder in 200OK SDP: ◉ No ○ Yes
Preferred Vocoder: (in listed order) choice 1: PCMU choice 2: G729 choice 3: PCMU choice 4: PCMU choice 5: PCMU choice 6: PCMU