Grandstream HT702 Configuration Guide for FreePhoneLine
Software Version: 1.0.6.1

Official FPL guidelines you can find at: http://support.freephoneline.ca/entries/23120323-VoIP-Unlock-Key-Credentials

All the settings were default on the device before configuration.

Step 1.
Plug all the appropriate wires to your Grandstream HT702 device.

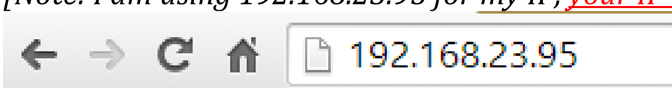
- Internet cable (Internet connection)
- Phone line (attached to the phone) *[Note: Use 'Port 1']*
- Power

Step 2.
From the phone attached to your Grandstream HT702 device:

- Dial '***'
- Then dial '02'

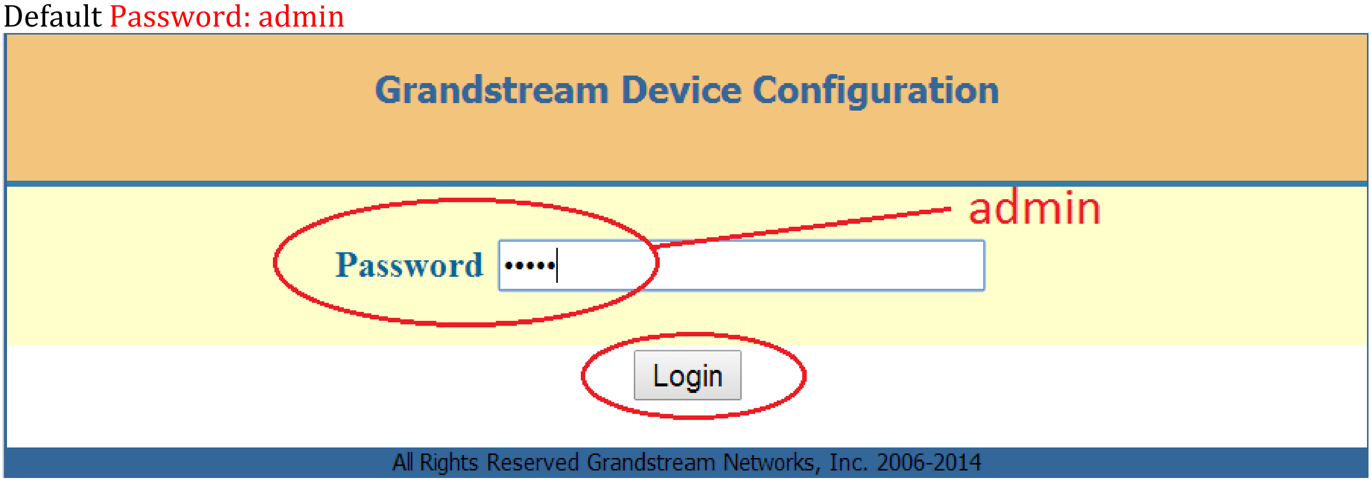
Write down your IP address.

Step 3.
On the PC/Mac connected to the same network as your Grandstream HT702 device, go to your browser (Internet Explorer; Chrome; Firefox; Opera; Safari, etc.).
In the URL bar put your IP address from Step 2.
[Note: I am using 192.168.23.95 for my IP, your IP address might be different!]

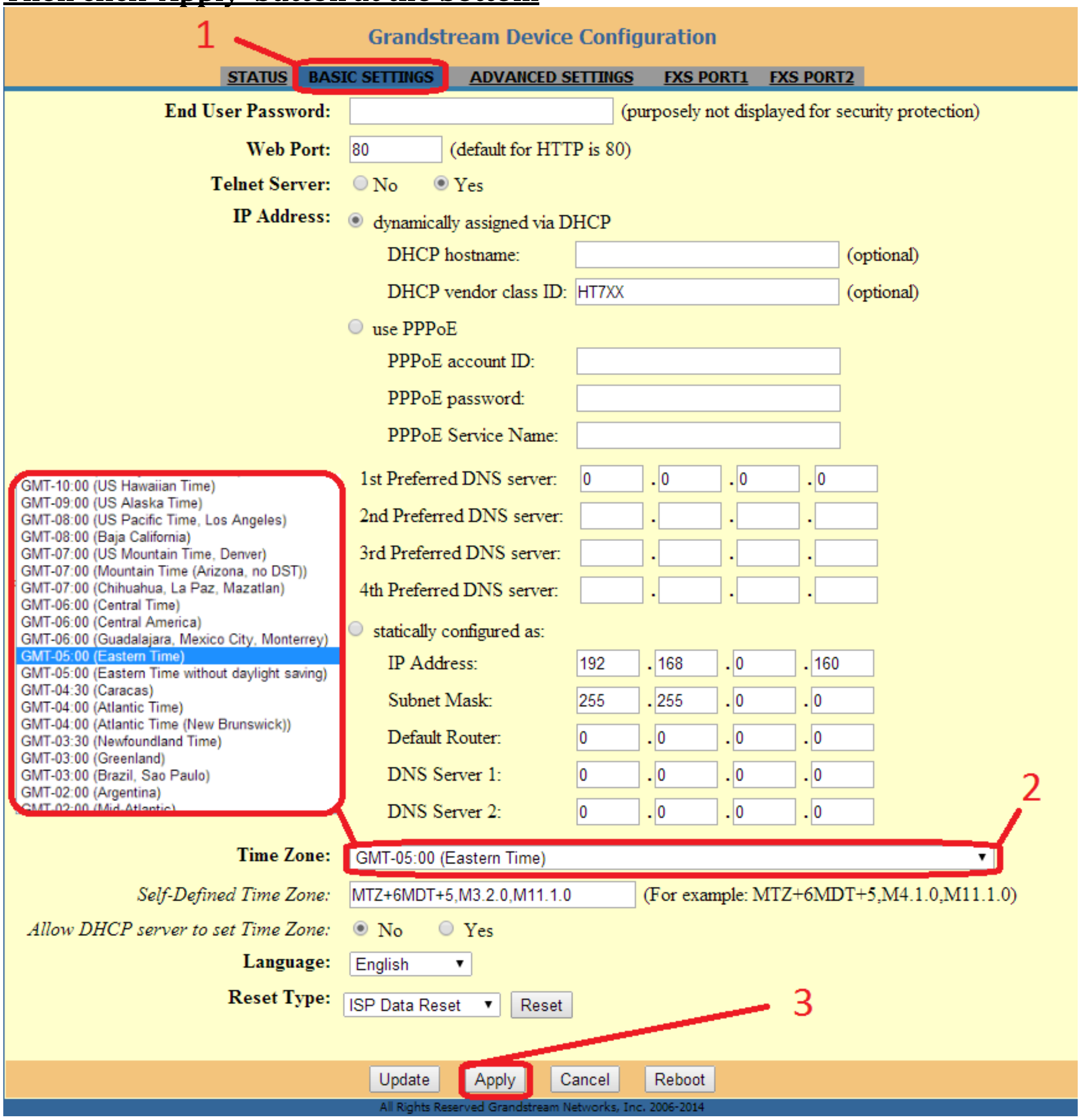


Web-based utility will appear.

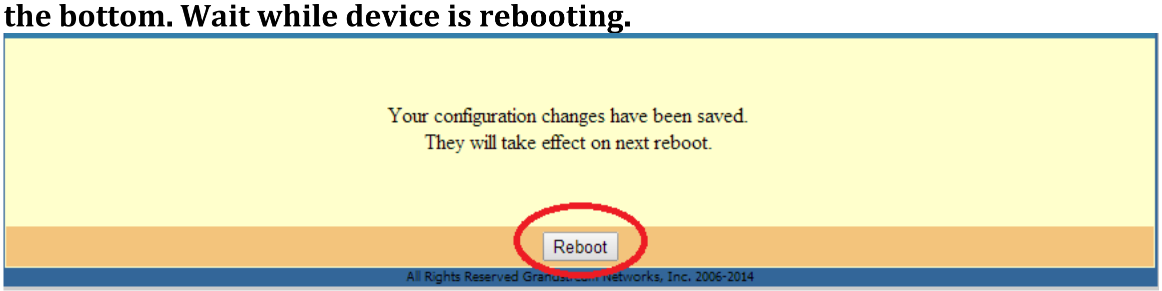
Now you have to login.
Default Password: admin



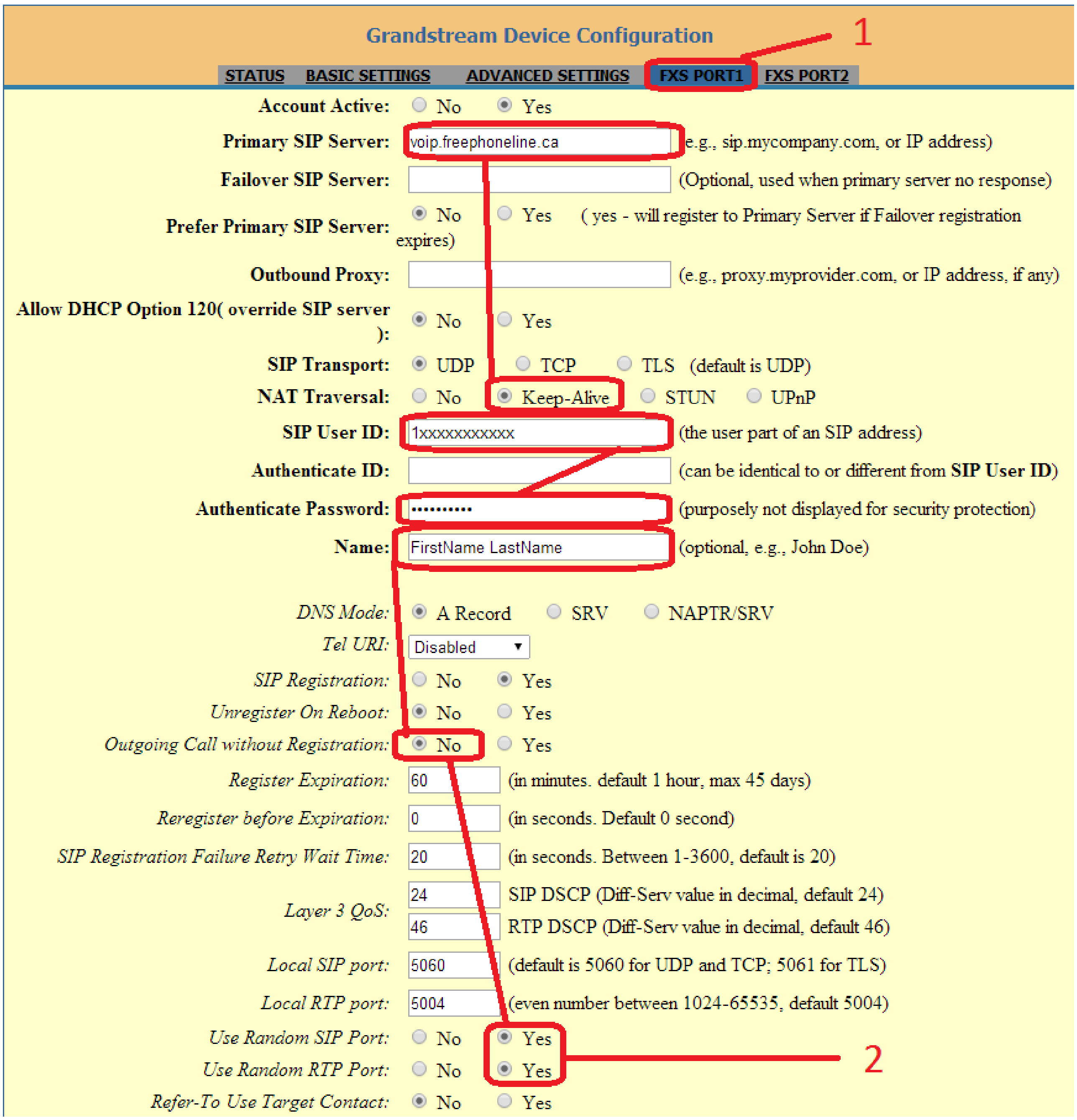
Step 4.
Go to 'BASIC SETTINGS' tab and fill in the following settings:
Time Zone: *[Choose appropriate time zone, depending where you are located]*
Then click 'Apply' button at the bottom



After you click 'Apply' button, device should ask you to reboot. If so – click on 'Reboot' button to apply the changes you have made. If it doesn't – go back to 'BASIC SETTINGS' tab and click 'Reboot' button at the bottom. Wait while device is rebooting.



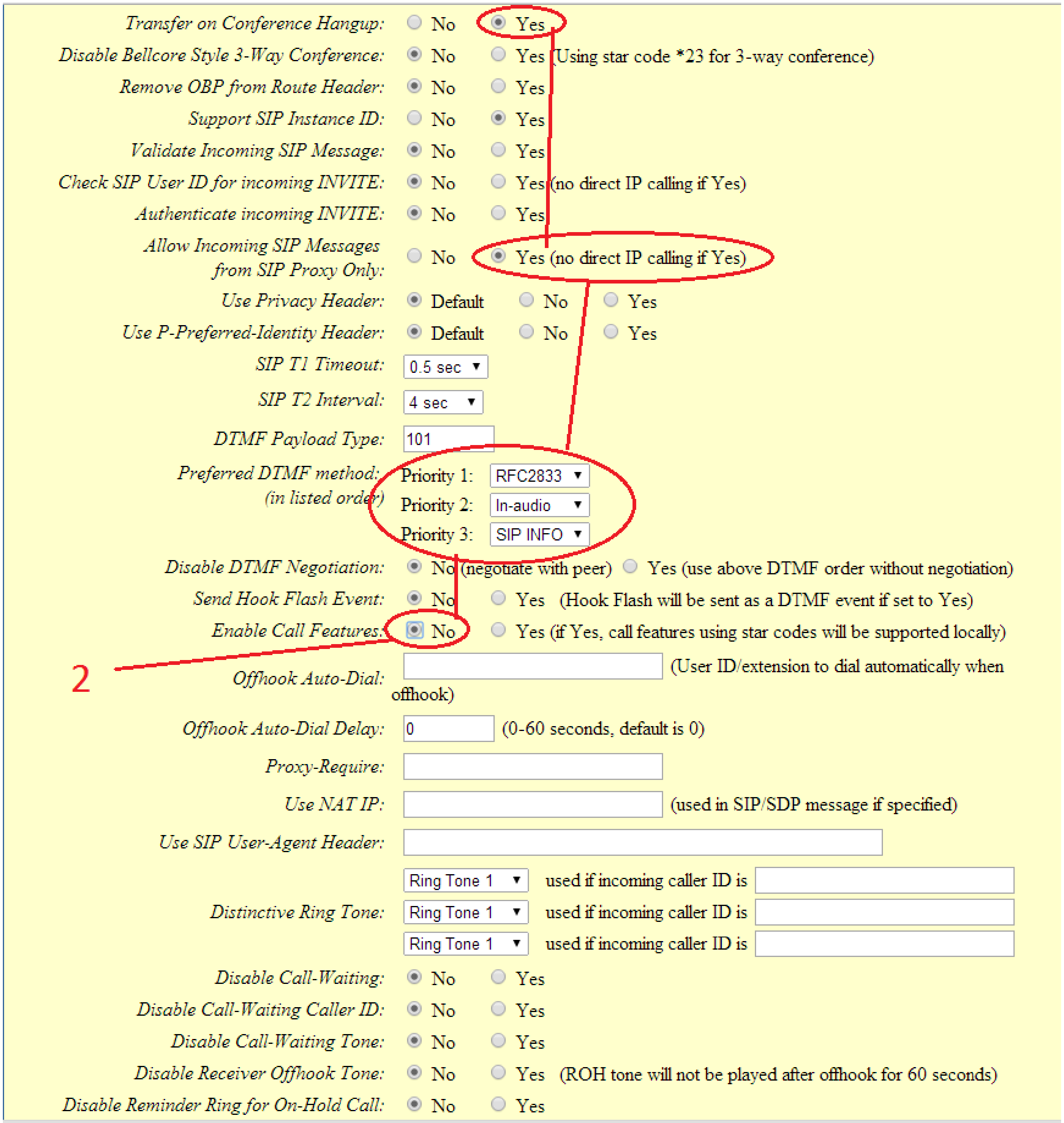
Step 5.
Go to 'FXS PORT 1' tab and change the following settings:
Primary SIP Server: *voip.freephoneline.ca* **OR** *voip2.freephoneline.ca* **[Note: For ROGERS Internet provider customers use *voip4.freephoneline.ca:6060*]**
NAT Traversal: *Keep-Alive*
SIP User ID: *[Your FPL number 1xxxxxxxxxx]*
Authenticate Password: *[Your SIP password]*
Name: *[Your first and last name]*
Outgoing Call without Registration: *No*
Use Random SIP Port: *Yes*
Use Random RTP Port: *Yes*



Transfer on Conference Hangup: *Yes*
Allow Incoming SIP Messages from SIP Proxy Only: *Yes*
Transfer on Conference Hangup: *Yes*
Preferred DTMF method: (in listing order)

Priority 1:	*RFC2833*
Priority 2:	*In-audio*
Priority 3:	*SIP INFO*

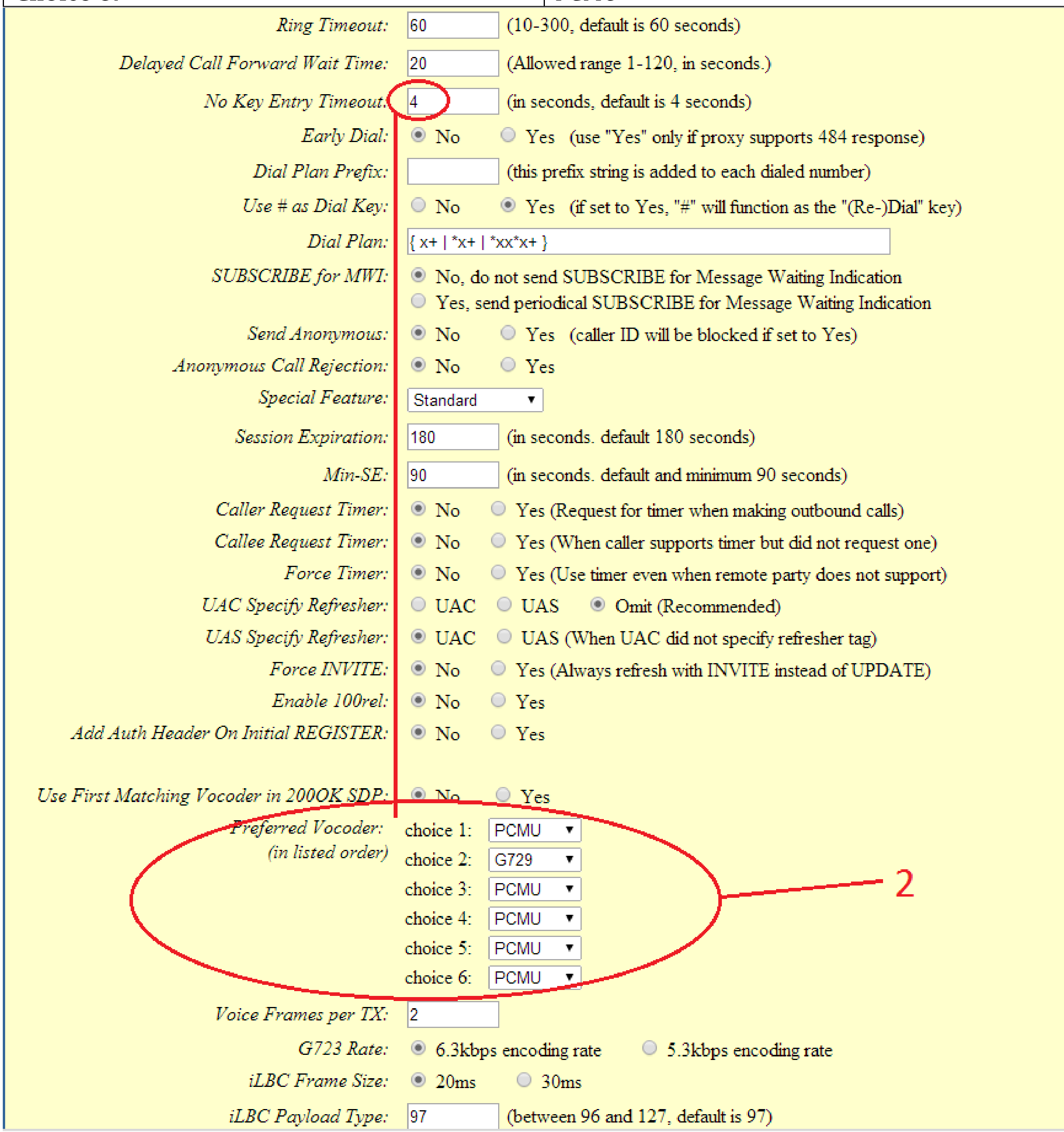
Enable Call Features: *No*



No Key Entry Timeout: *4*

Preferred Vocoder: (in listed order):

Choice 1:	*PCMU*
Choice 2:	*G729*
Choice 3:	*PCMU*
Choice 4:	*PCMU*
Choice 5:	*PCMU*
Choice 6:	*PCMU*



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:	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: G729 choice 3: PCMU choice 4: PCMU choice 5: PCMU choice 6: PCMU
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)

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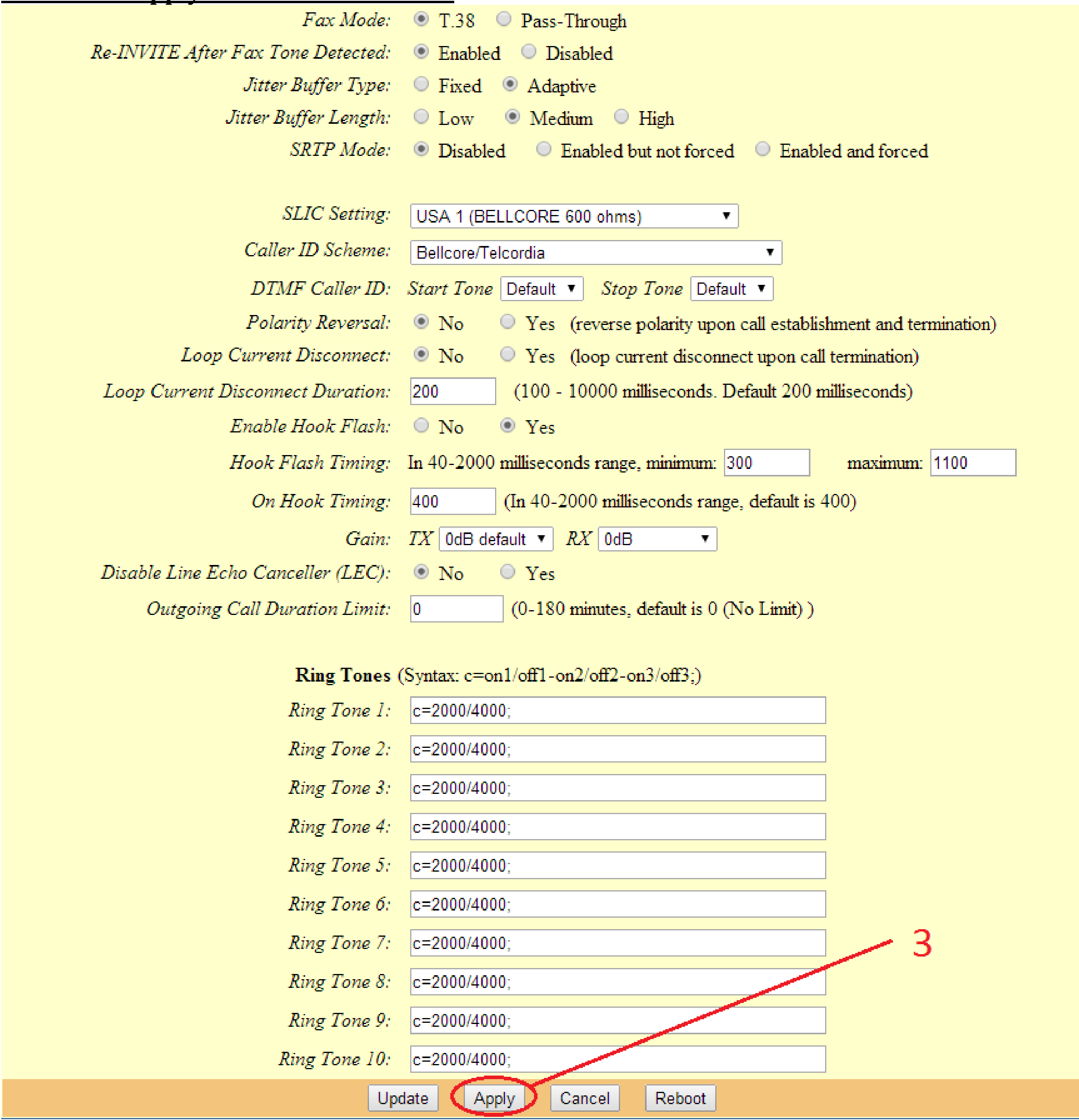
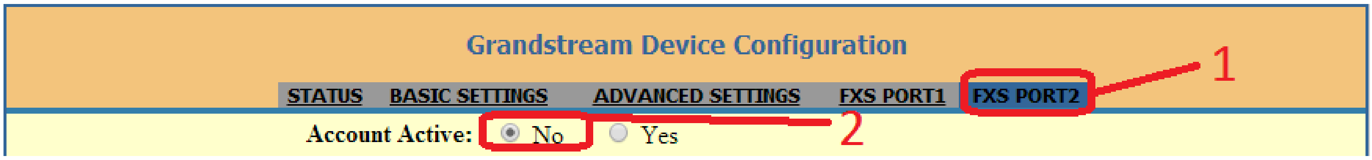
Then click 'Apply' button at the bottom

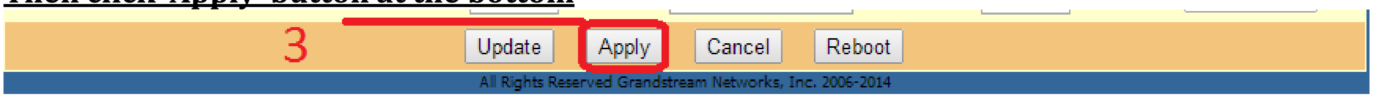


The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT 1' tab and click 'Reboot' at the bottom. Wait while device is rebooting.

Step 6.
Go to 'FXS PORT 2' tab and change the following settings:
Account Active: *No*



Then click 'Apply' button at the bottom



The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT 2' tab and click 'Reboot' at the bottom. Wait while device is rebooting.

Enjoy your free phone line! ☺