Grandstream HT701 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 HT701 device.

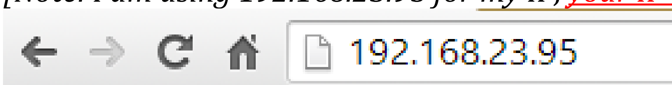
- Internet cable (Internet connection)
- Phone line (attached to the phone)
- Power

Step 2.
From the phone attached to your Grandstream HT701 device:

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

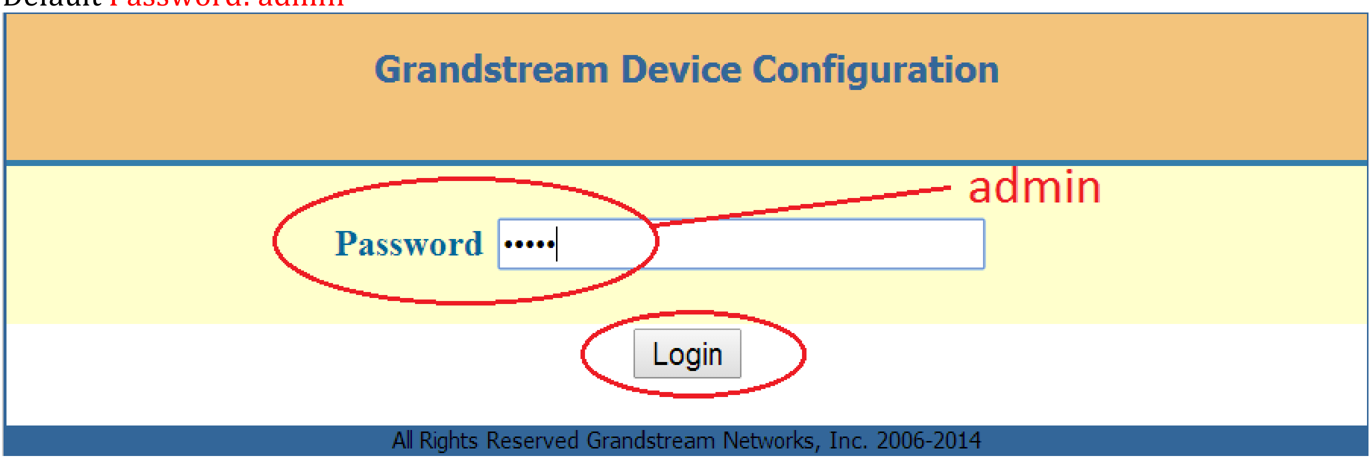
Write down your IP address.

Step 3.
On the PC/Mac connected to the same network as your Grandstream HT701 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.

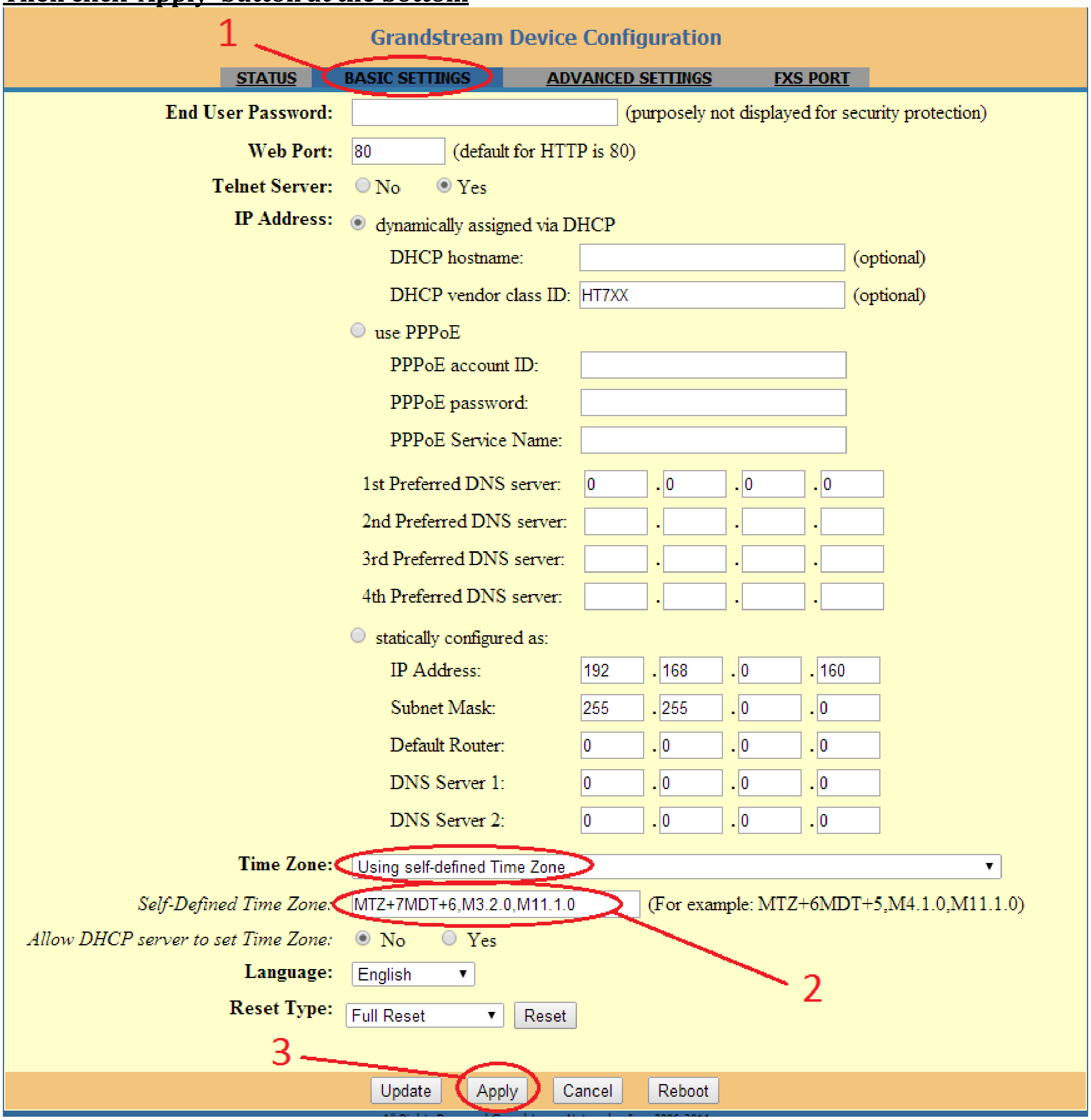
Now you have to login.
Default Password: admin



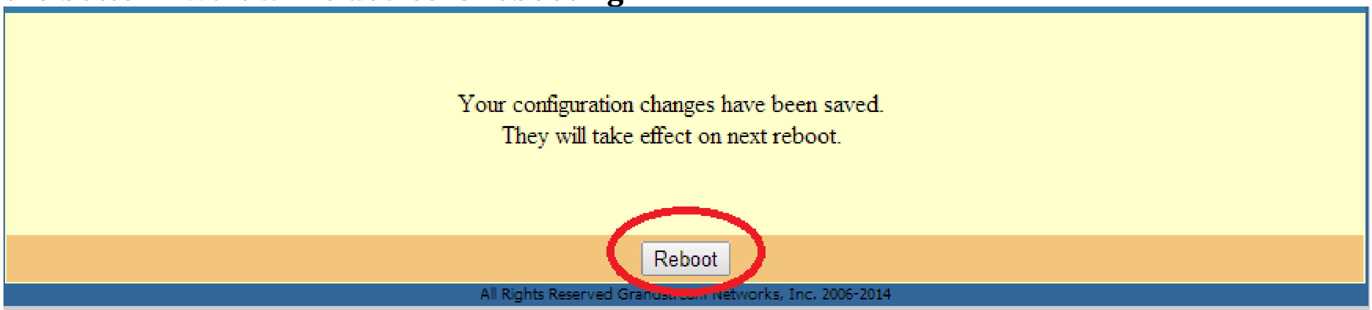
Step 4.
Go to 'BASIC SETTINGS' tab and fill in the following settings:
Time Zone: *Using self-defined Time Zone*
Self-Defined Time Zone: [Choose appropriate time zone, depending where you are located]

Your time zone	**What value to fill in**
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*

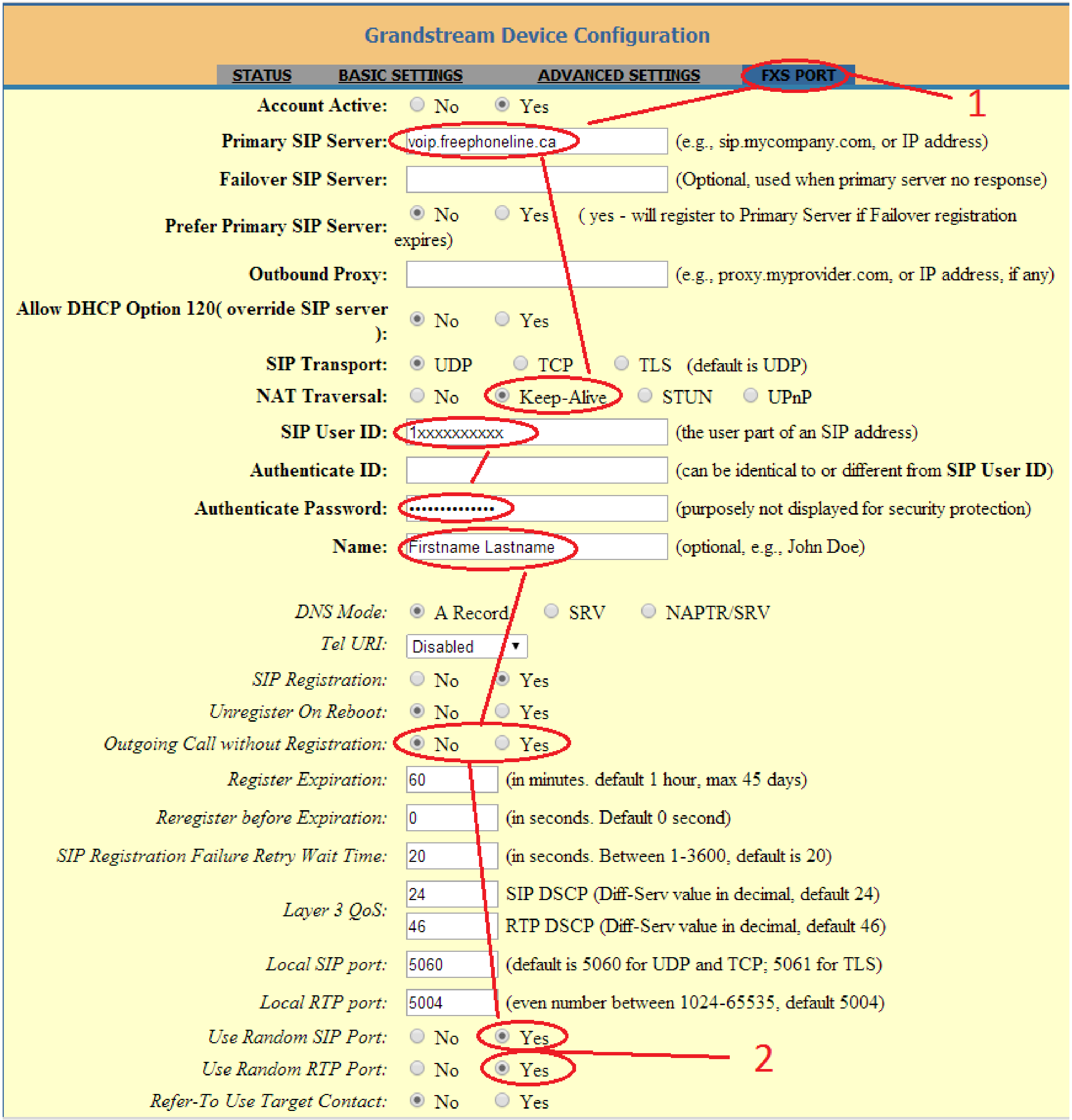
Then click 'Apply' button at the bottom



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' 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*



Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | FXS PORT

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Account Active:	○ No ◉ Yes	
Primary SIP Server:	voip.freephoneline.ca	(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:	1xxxxxxxxxx	(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:	Firstname Lastname	(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)
Layer 3 QoS:	24	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	

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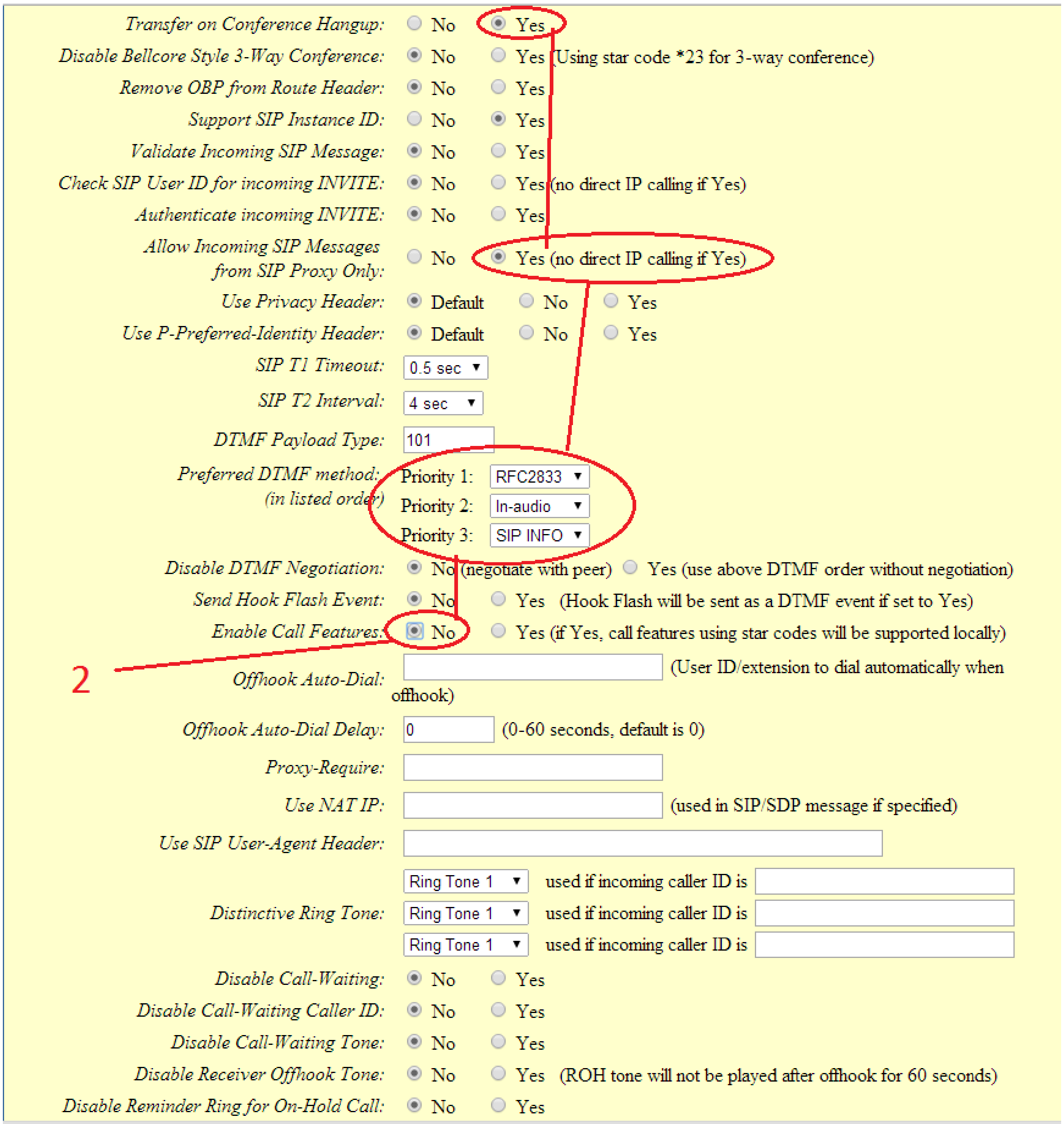
Transfer on Conference Hangup: *Yes*
Allow Incoming SIP Messages from SIP Proxy Only: *Yes*
Transfer on Conference Hangup: *Yes*
Preferred DTMF method: (in listing order)

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

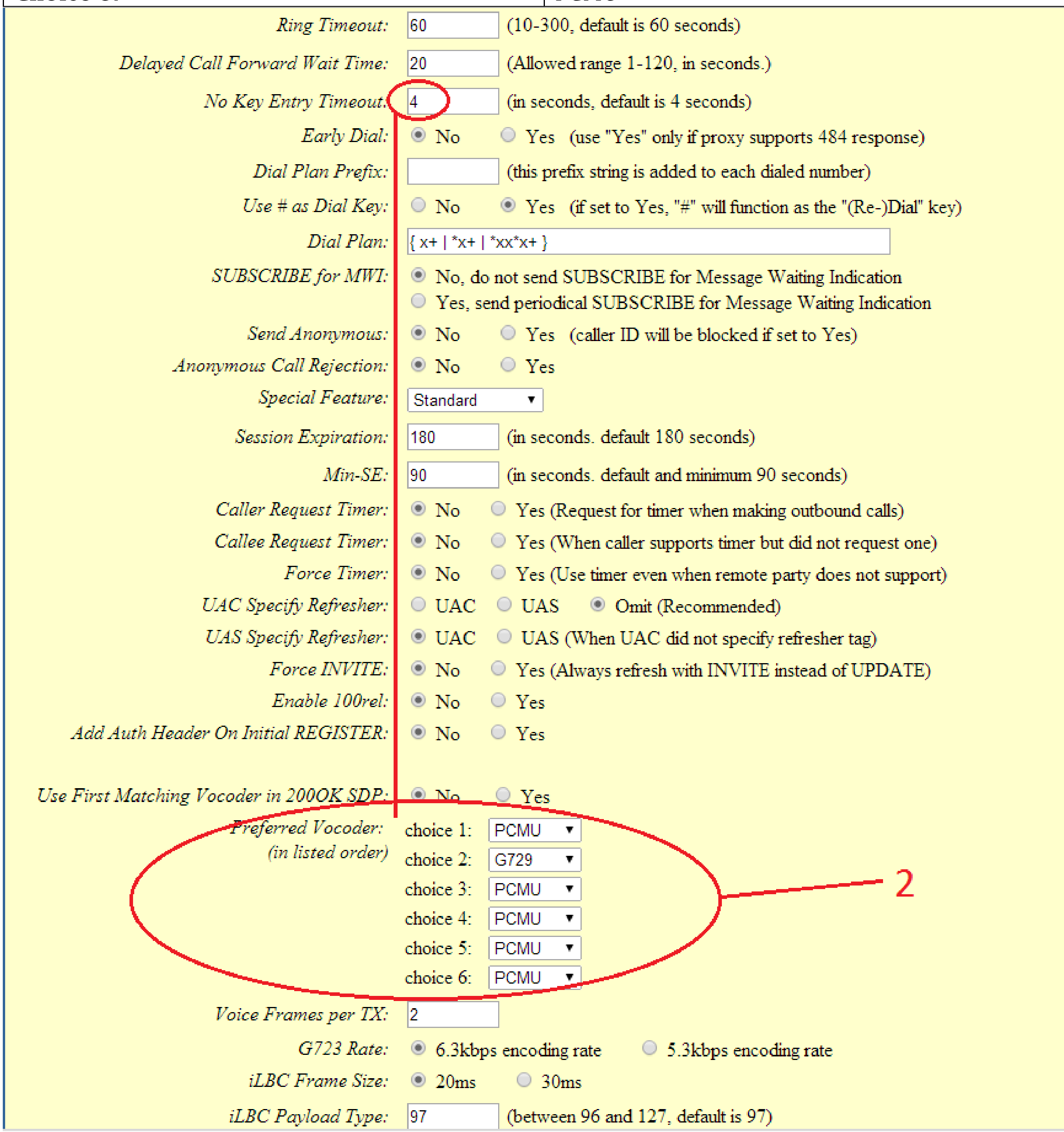
Enable Call Features: *No*



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
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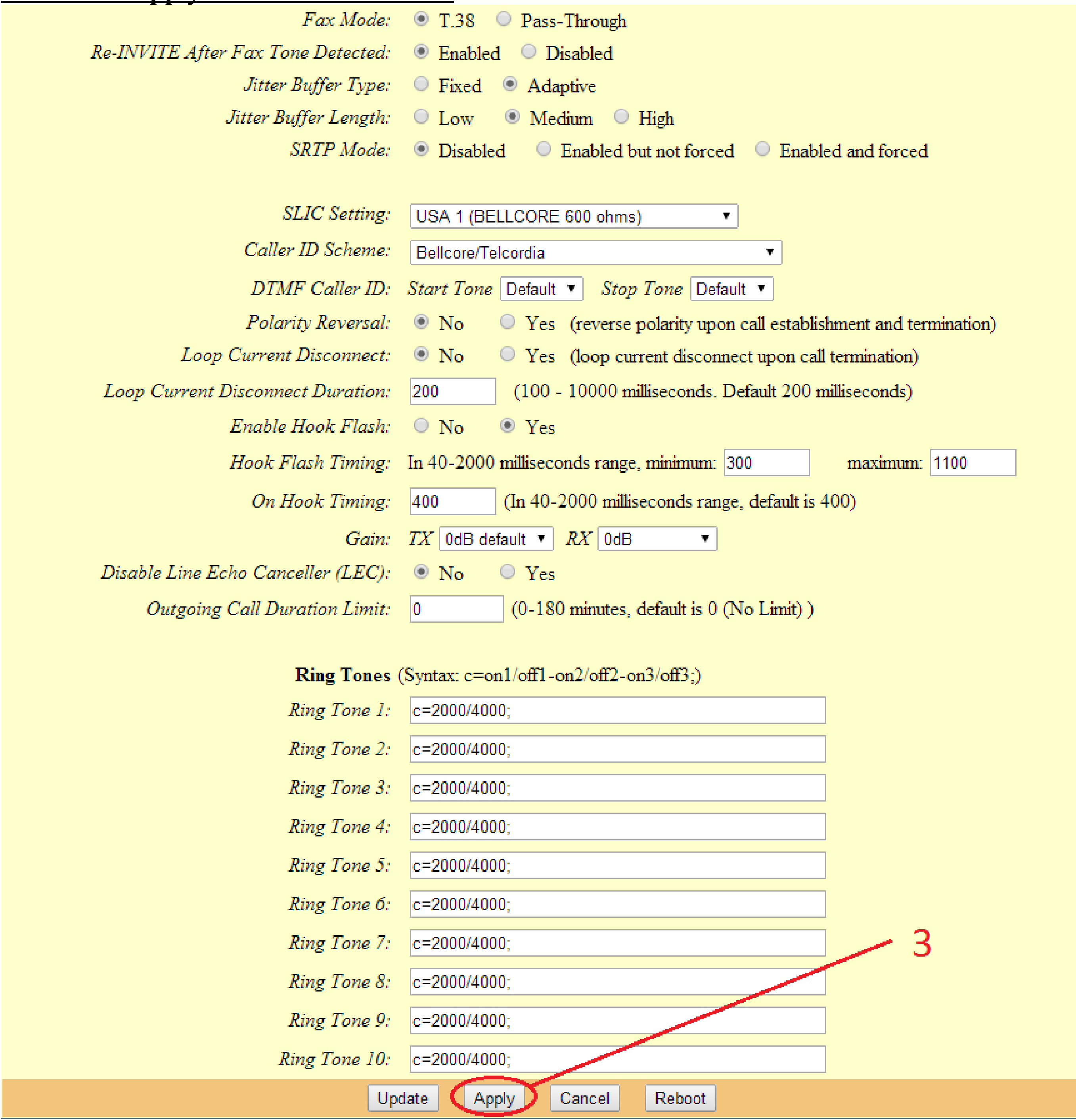
Voice Frames per TX: 2
G723 Rate: 6.3kbps encoding rate 5.3kbps encoding rate
iLBC Frame Size: 20ms 30ms
iLBC Payload Type: 97 (between 96 and 127, default is 97)

<u>Then click 'Apply' button at the bottom</u>



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

Enjoy your free phone line! ☺