Linksys PAP2T NA Configuration Guide for FreePhoneLine

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 Linksys PAP2T device.

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

Step 2.

You have to figure out IP address to access web-based utility. From your phone dial:

- '****'
- Then dial '110#'

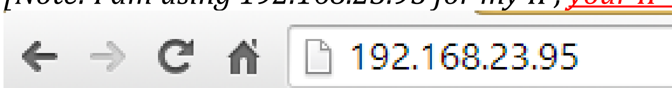
Write down your IP address.

Step 3.

On the PC/Mac connected to the same network as your Linksys PAP2T device, go to your browser (Internet Explorer; Chrome; Firefox; Opera; Safari, etc.).

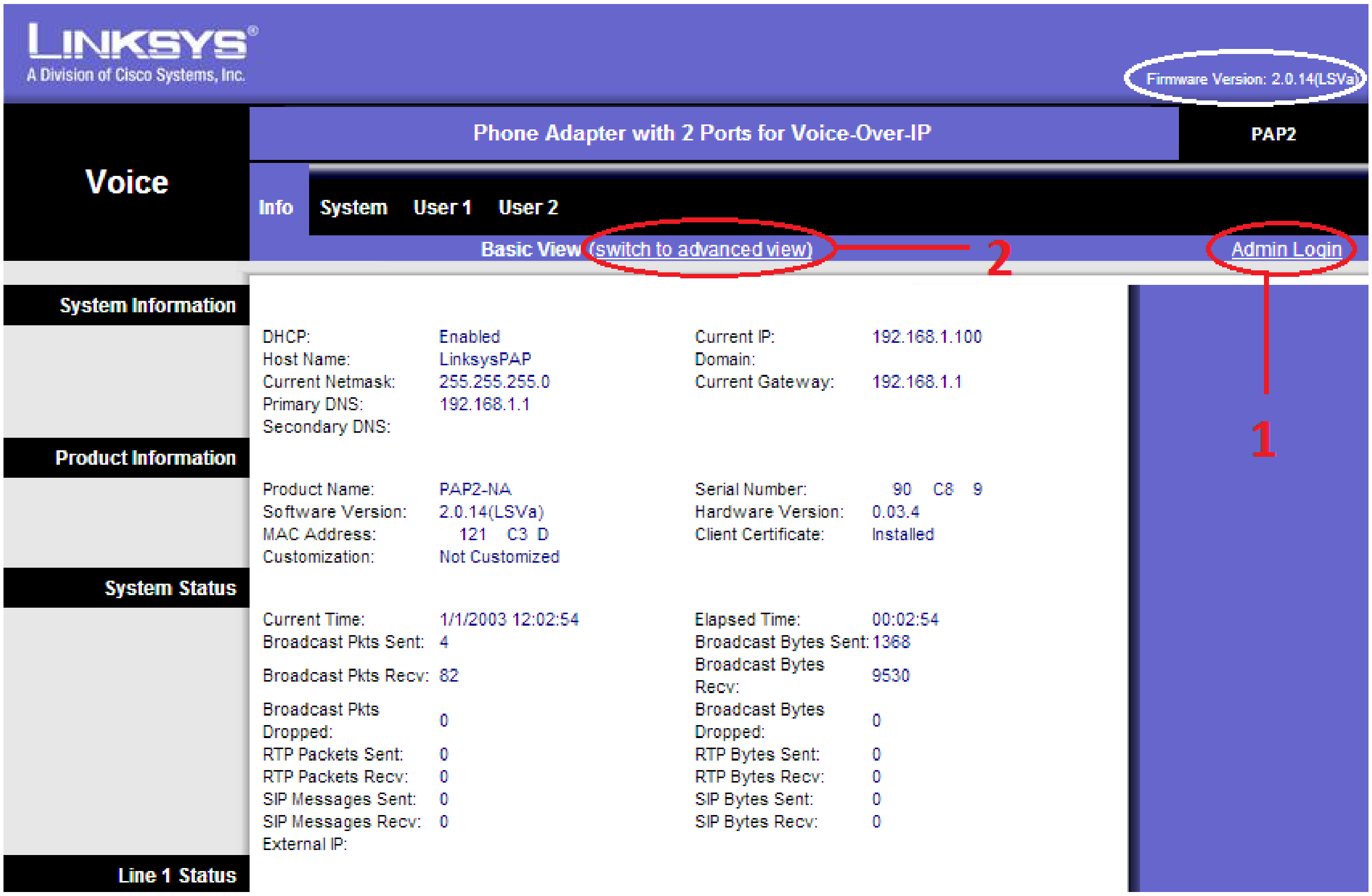
In the URL bar put your IP address from Step 2.

[Note: I am using 192.168.23.95 for my IP, your IP address might be different!]

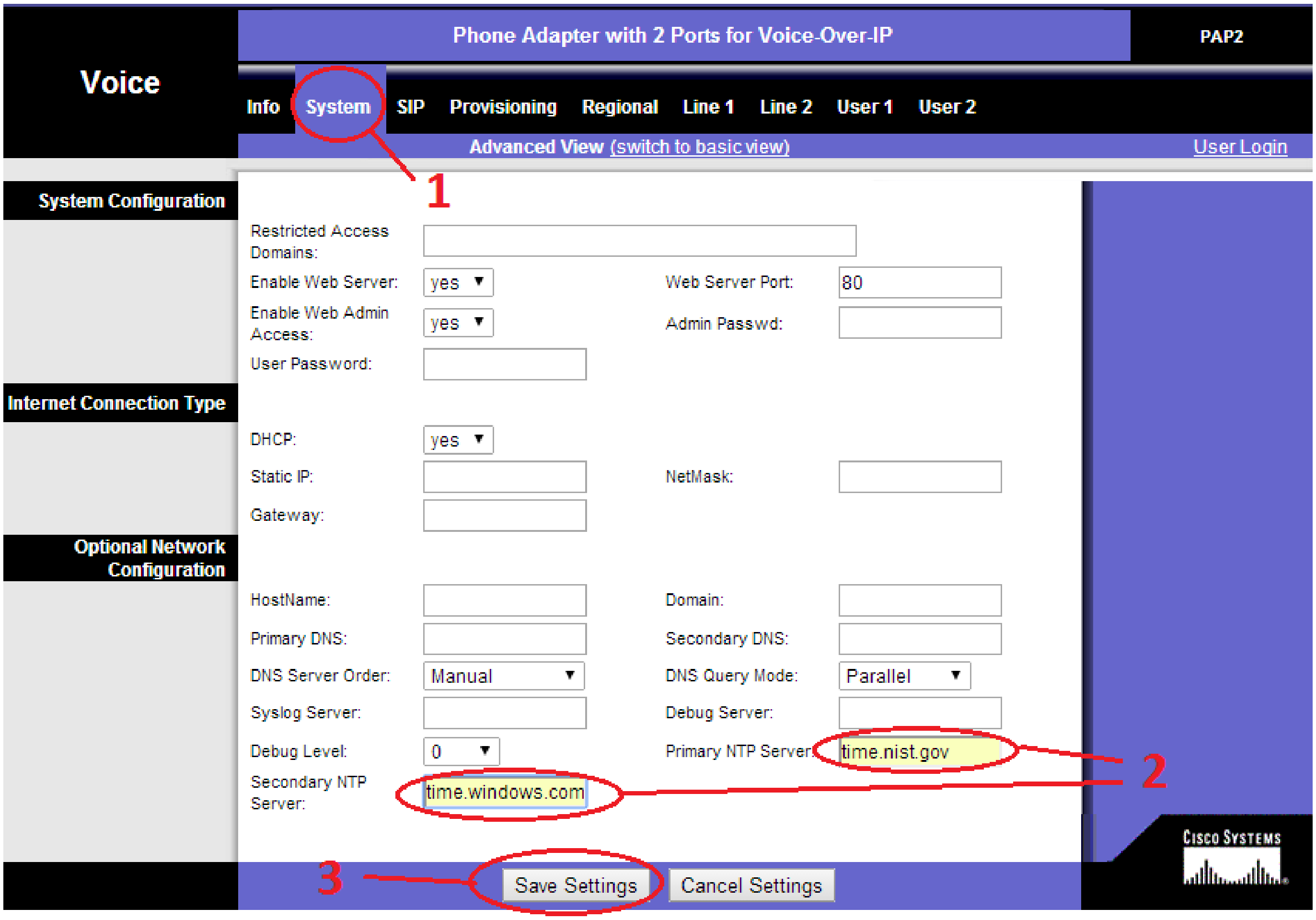


Web-based utility will appear.

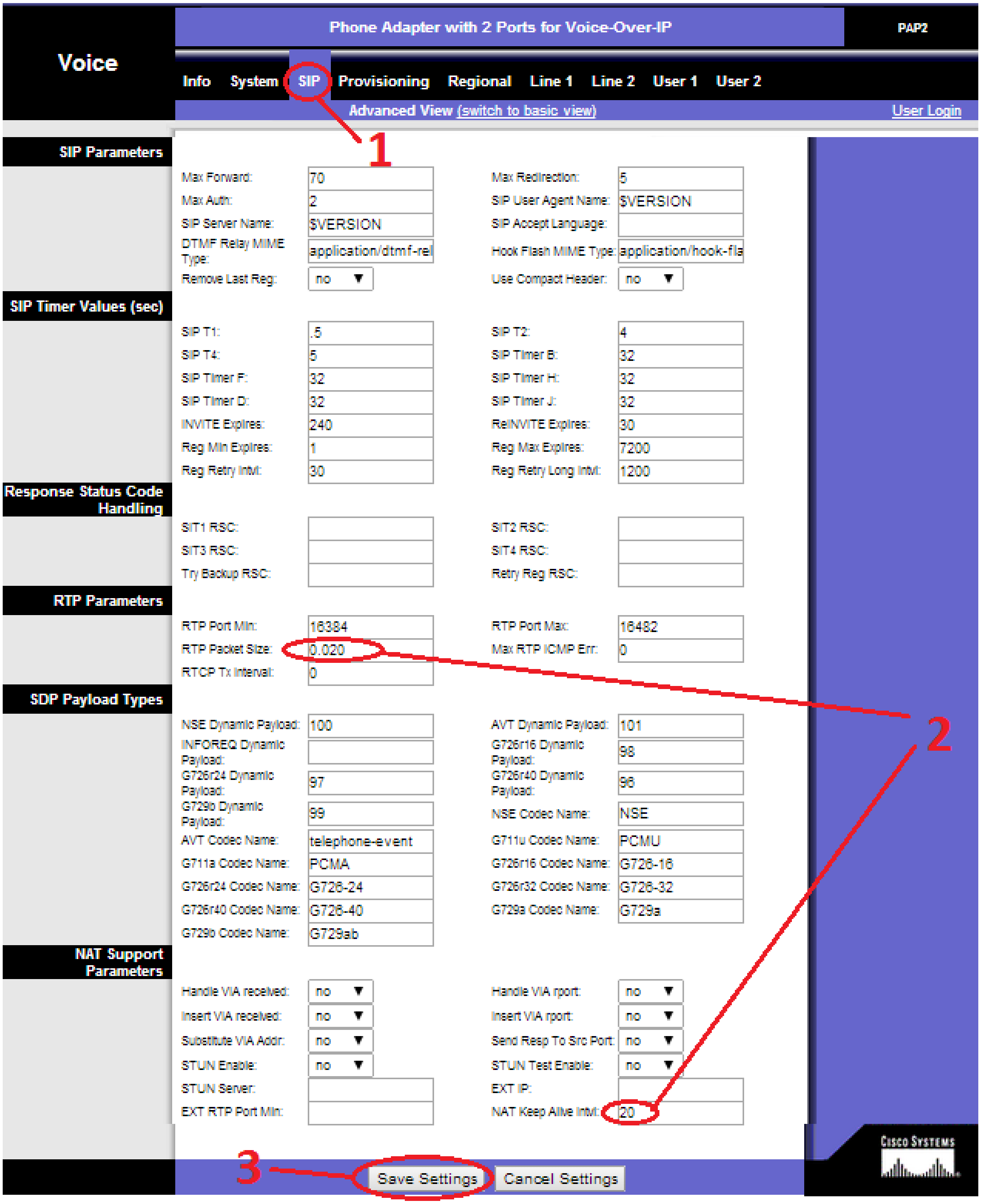
You have to click 'Admin Login', then switch from 'basic' to 'advanced view'.



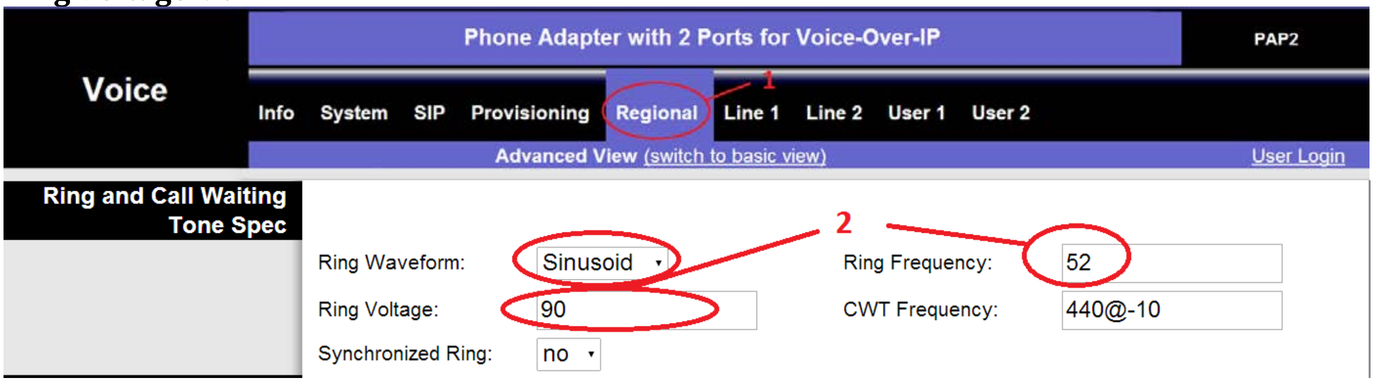
Step 4.
On the ‘System’ tab fill in the following settings:
Under ‘Optional Network Configuration’:
Primary NTP Server: *time.nist.gov*
Secondary NTP Server: *time.windows.com*
Then click ‘Save Settings’ button at the bottom.
Wait until changes are saved.



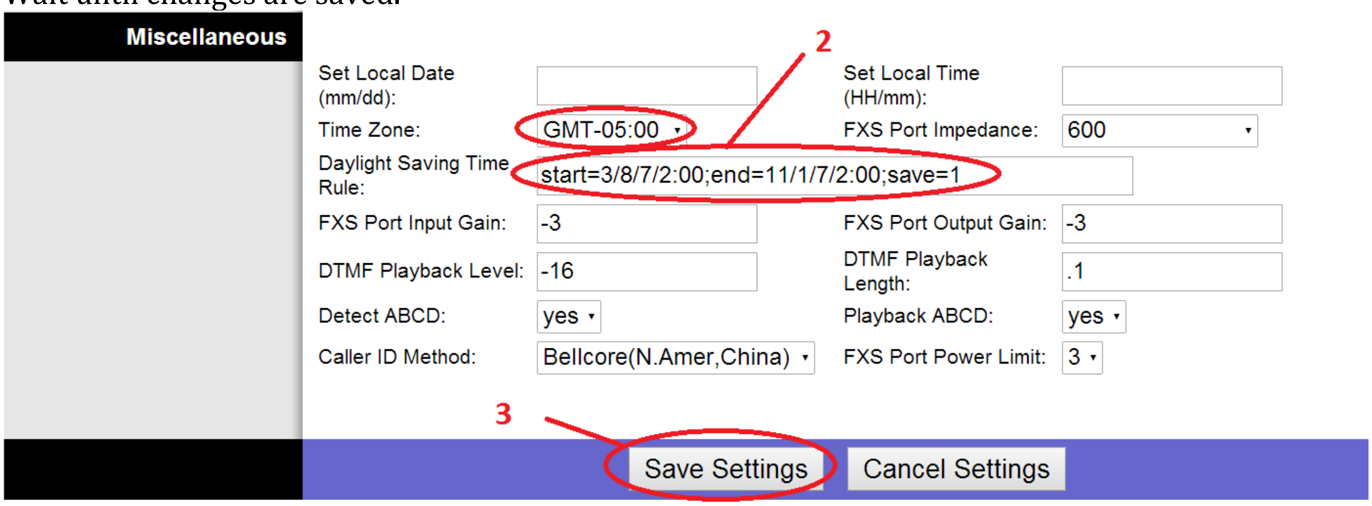
<u>Step 5.</u>
Go to ‘SIP’ tab and change the following settings:
Under ‘RTP Parameters’ category:
RTP Packet Size: *0.020*
Under ‘NAT Support Parameters’ category:
NAT Keep Alive Intvl: *20*
Then click ‘Save Settings’ button at the bottom.
Wait until changes are saved.


Voice
Phone Adapter with 2 Ports for Voice-Over-IP
PAP2
Info System SIP Provisioning Regional Line 1 Line 2 User 1 User 2
Advanced View (switch to basic view)
User Login
1
SIP Parameters
Max Forward: 70
Max Redirection: 5
Max Auth: 2
SIP User Agent Name: $VERSION
SIP Server Name: $VERSION
SIP Accept Language:
DTMF Relay MIME Type: application/dtmf-rel
Hook Flash MIME Type: application/hook-fla
Remove Last Reg: no
Use Compact Header: no
SIP Timer Values (sec)
SIP T1: .5
SIP T2: 4
SIP T4: 5
SIP Timer B: 32
SIP Timer F: 32
SIP Timer H: 32
SIP Timer D: 32
SIP Timer J: 32
INVITE Expires: 240
ReINVITE Expires: 30
Reg Min Expires: 1
Reg Max Expires: 7200
Reg Retry Intvl: 30
Reg Retry Long Intvl: 1200
Response Status Code Handling
SIT1 RSC:
SIT2 RSC:
SIT3 RSC:
SIT4 RSC:
Try Backup RSC:
Retry Reg RSC:
RTP Parameters
RTP Port Min: 16384
RTP Port Max: 16482
RTP Packet Size: 0.020
Max RTP ICMP Err: 0
RTCP Tx Interval: 0
2
SDP Payload Types
NSE Dynamic Payload: 100
AVT Dynamic Payload: 101
INFOREQ Dynamic Payload:
G726r16 Dynamic Payload: 98
G726r24 Dynamic Payload: 97
G726r40 Dynamic Payload: 96
G729b Dynamic Payload: 99
NSE Codec Name: NSE
AVT Codec Name: telephone-event
G711u Codec Name: PCMU
G711a Codec Name: PCMA
G726r16 Codec Name: G726-16
G726r24 Codec Name: G726-24
G726r32 Codec Name: G726-32
G726r40 Codec Name: G726-40
G729a Codec Name: G729a
G729b Codec Name: G729ab
NAT Support Parameters
Handle VIA received: no
Handle VIA rport: no
Insert VIA received: no
Insert VIA rport: no
Substitute VIA Addr: no
Send Resp To Src Port: no
STUN Enable: no
STUN Test Enable: no
STUN Server:
EXT IP:
EXT RTP Port Min:
NAT Keep Alive Intvl: 20
3
Save Settings Cancel Settings
Cisco Systems

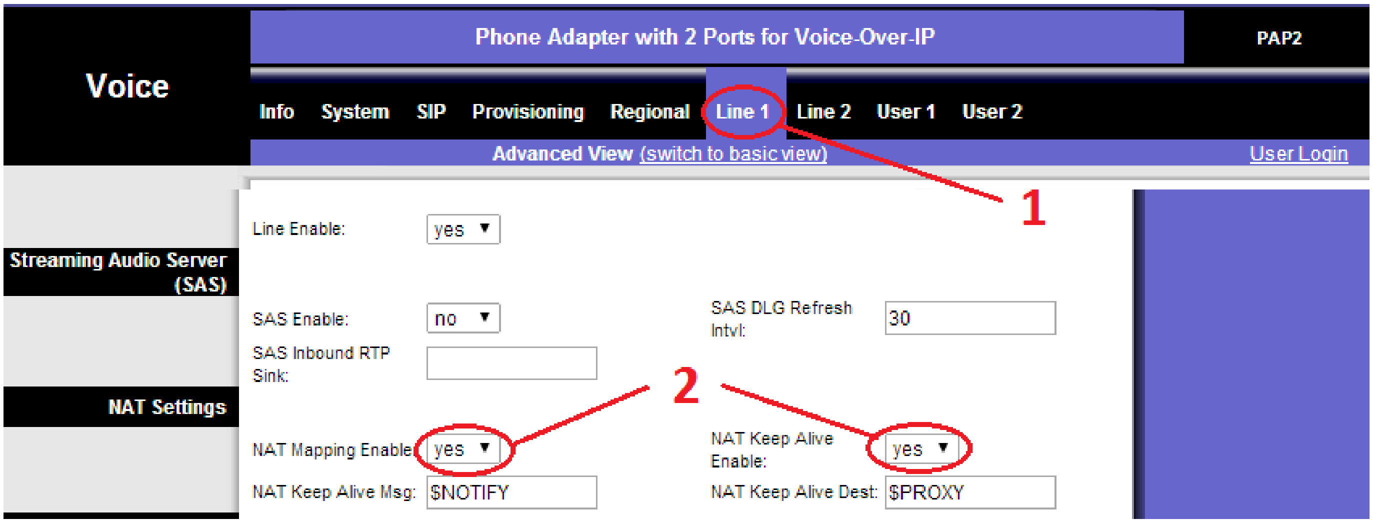
<u>Step 6.</u>
Go to 'Regional' tab and change the following settings:
Under 'Ring and Call Waiting Tone Spec' category:
Ring Waveform: *Sinusoid*
Ring Frequency: *52*
Ring Voltage: *90*



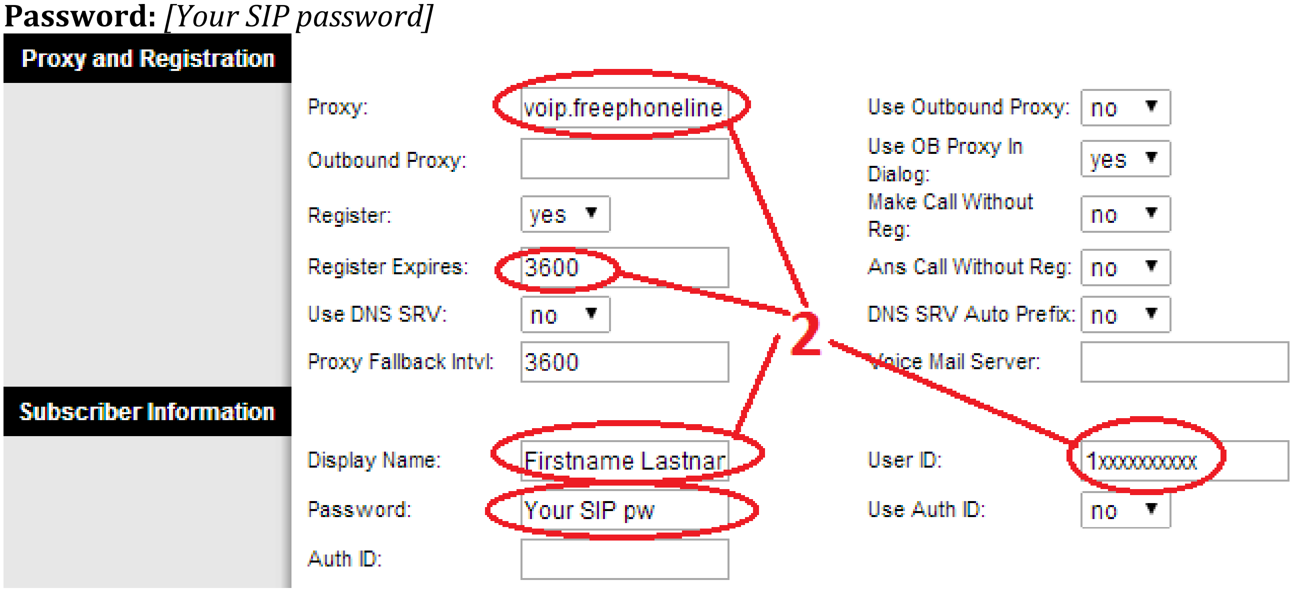
Under 'Miscellaneous' category:
Time Zone: *GMT-5*
Daylight Savings Time Rule: *start=3/8/7/2:00;end=11/1/7/2:00;save=1*
Then click 'Submit All Changes' button at the bottom.
Wait until changes are saved.



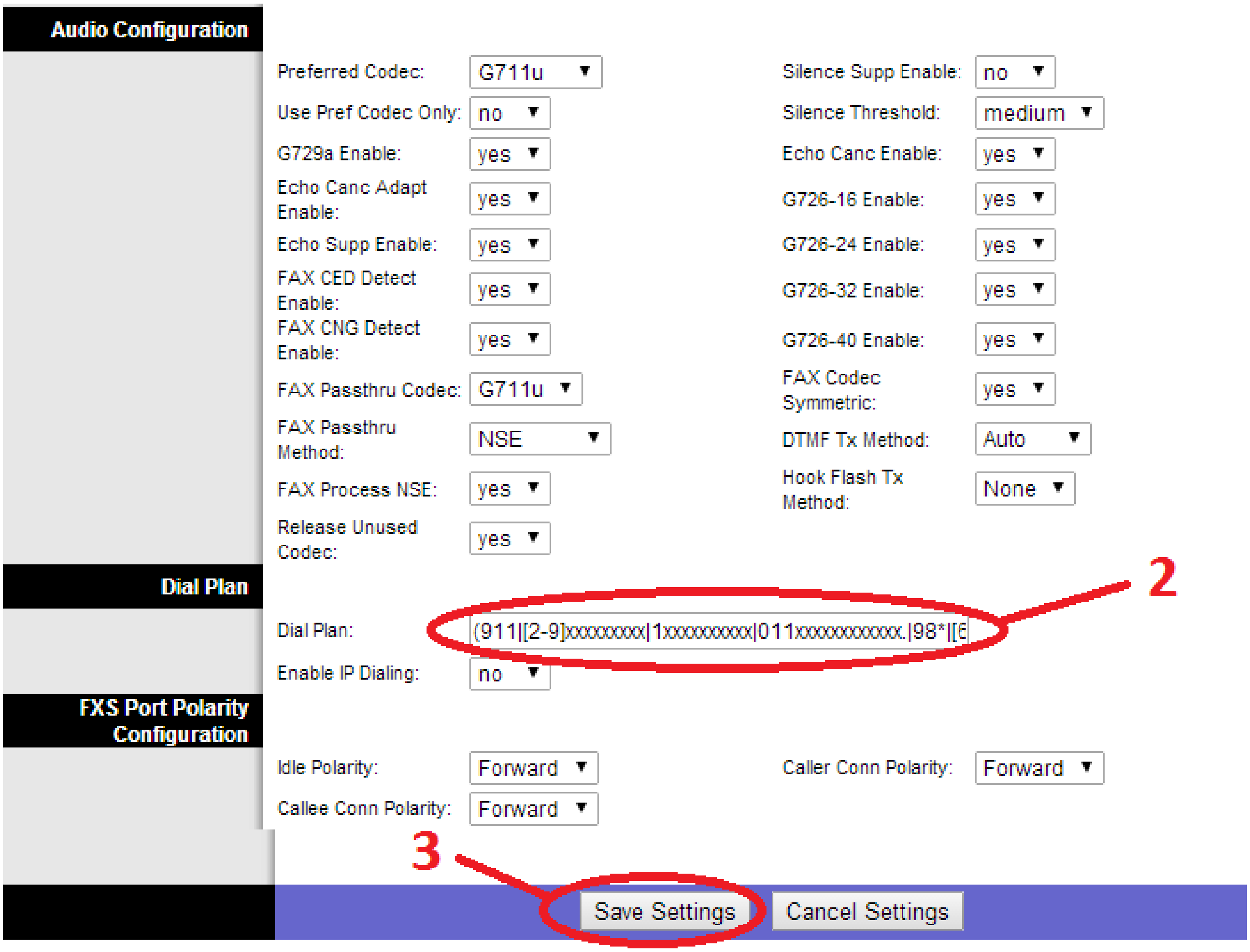
<u>Step 7.</u>
Go to 'Line 1' tab and change the following settings:
Under 'NAT Settings' category:
NAT Mapping Enable: *yes*
NAT Keep Alive Enable: yes



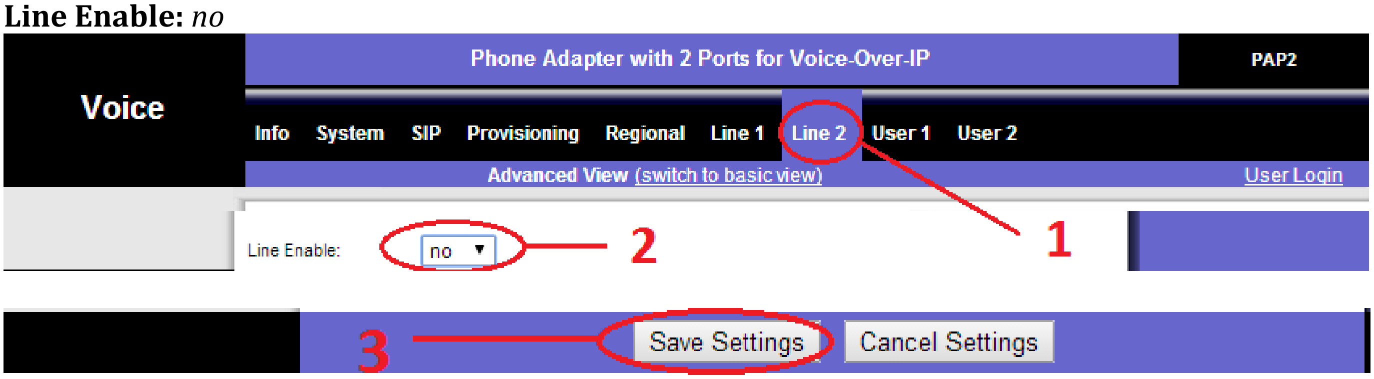
Under 'Proxy and Registration' category:
Proxy: *voip.freephoneline.ca* **OR** *voip2.freephoneline.ca* **[Note: For ROGERS Internet provider customers use *voip4.freephoneline.ca:6060*]**
Register Expires: *3600*
Under 'Subscriber Information' category:
Display Name: *[Your first and last name]*
[Note: ATA and SIP clients with a Caller ID string containing non-alphanumeric characters will prevent you from making outgoing calls]
User ID: *[Your FPL number 1xxxxxxxxxx]*
Password: *[Your SIP password]*



Under 'Dial Plan' category:
Dial Plan: *(911|[2-9]xxxxxxxxx|1xxxxxxxxxx|011xxxxxxxxxxx.|98*|[6-7]x*xxxxxxxxxx.)*
Then click 'Save Settings' button at the bottom.
Wait until changes are saved.



<u>Step 8.</u>
Go to 'Voice' tab, then go to 'Line 2' tab:
Line Enable: *no*



<u>Then click 'Save Settings' button at the bottom.</u>
Wait until changes are saved.

Enjoy your free phone line! ☺