Linksys SPA3102 Configuration Guide for FreePhoneLine
Software Version: 5.2.13

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.

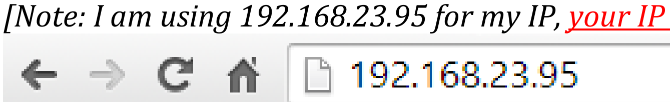
Step 1.
Plug all the appropriate wires to your Linksys SPA3102 device.
- Internet cable (Internet connection) *[Note: Use 'Internet' port]*
- Phone line (attached to the phone) *[Note: Use 'Phone' port]*
- Power

Step 2.
By default settings Linksys SPA3102 WAN access is disabled.
You have to enable this setting before you access web-based utility.
From the phone attached to your Linksys SPA3102 device:
- Dial '****'
- Then dial '7932#'
- Then press '1#' (to enable WAN access)
- Then press '1' (to save changes)
- Hang up the phone

Now, we have to figure out IP address to access web-based utility. From your phone dial:
- '****'
- Then dial '110#'

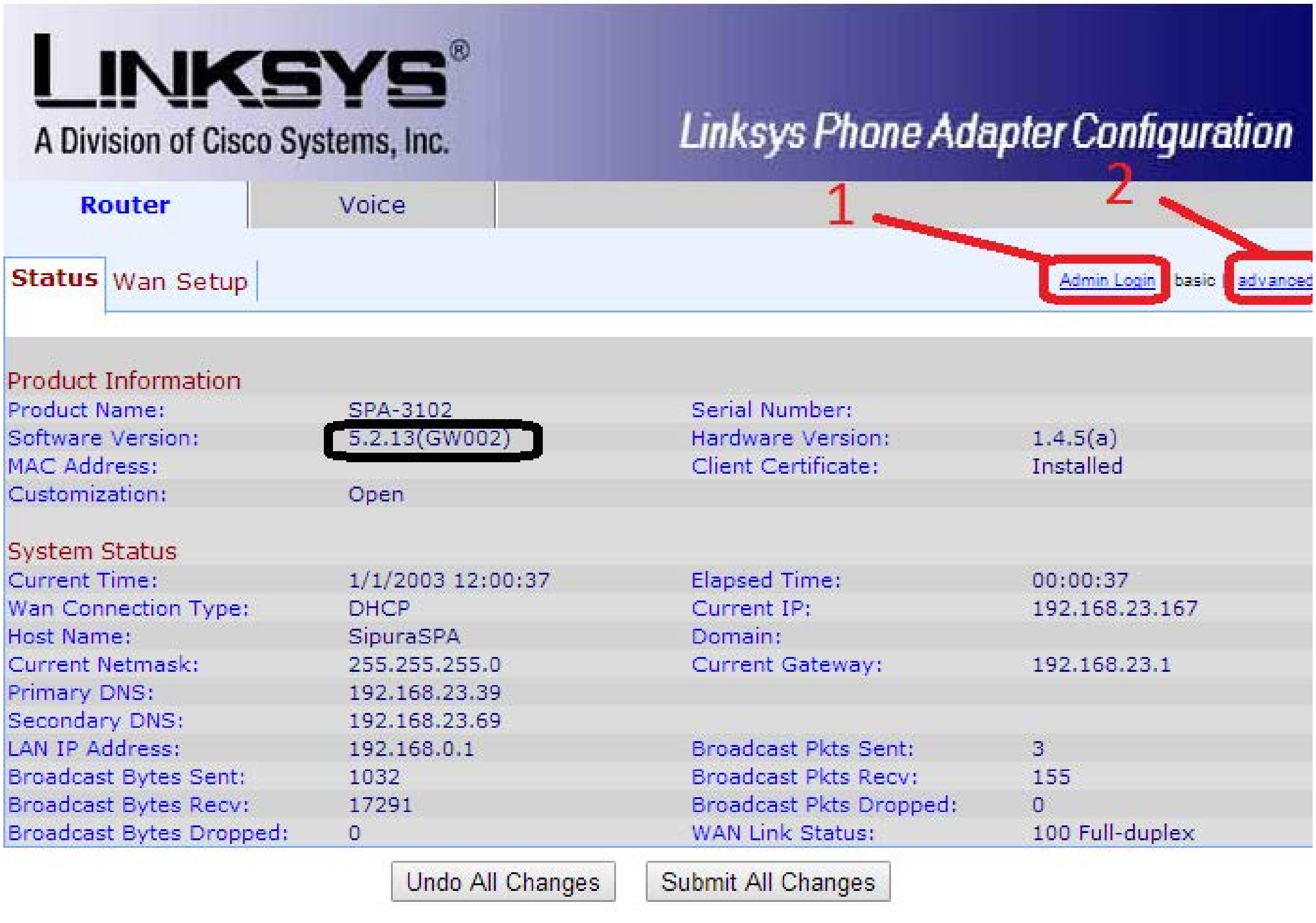
Write down your IP address.

Step 3.
On the PC/Mac connected to the same network as your Linksys SPA3102 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'.


LINKSYS®
A Division of Cisco Systems, Inc.
Linksys Phone Adapter Configuration
Router
Voice
1
2
Admin Login
basic
advanced
Status
Wan Setup
Product Information
Product Name: SPA-3102
Serial Number:
Software Version: 5.2.13(GW002)
Hardware Version: 1.4.5(a)
MAC Address:
Client Certificate: Installed
Customization: Open
System Status
Current Time: 1/1/2003 12:00:37
Elapsed Time: 00:00:37
Wan Connection Type: DHCP
Current IP: 192.168.23.167
Host Name: SipuraSPA
Domain:
Current Netmask: 255.255.255.0
Current Gateway: 192.168.23.1
Primary DNS: 192.168.23.39
Secondary DNS: 192.168.23.69
LAN IP Address: 192.168.0.1
Broadcast Pkts Sent: 3
Broadcast Bytes Sent: 1032
Broadcast Pkts Recv: 155
Broadcast Bytes Recv: 17291
Broadcast Pkts Dropped: 0
Broadcast Bytes Dropped: 0
WAN Link Status: 100 Full-duplex
Undo All Changes
Submit All Changes

;Step 4.

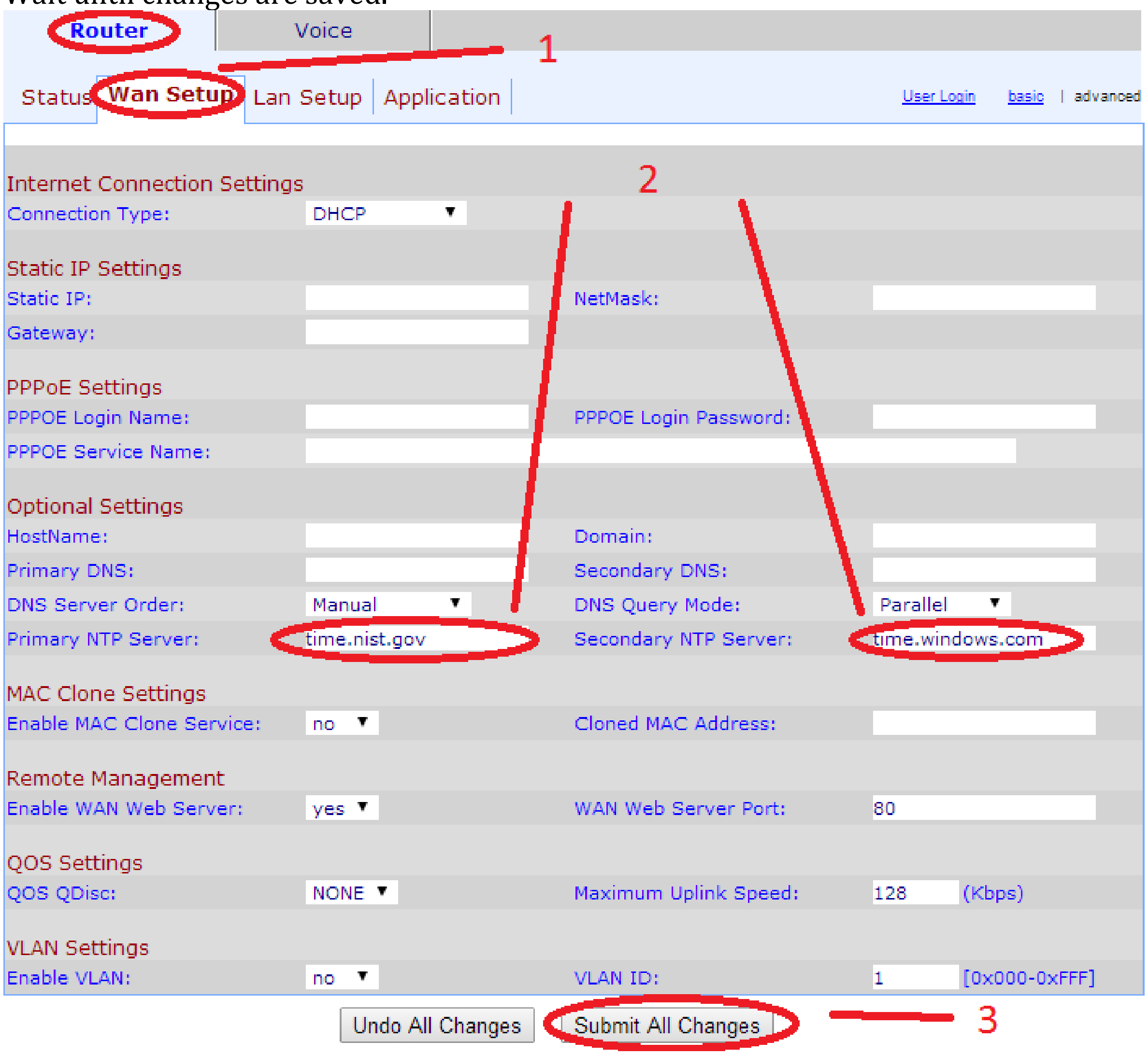
On the ‘Router’ tab go to ‘Wan Setup’ tab and fill in the following settings:

Primary NTP Server: *time.nist.gov*

Secondary NTP Server: *time.windows.com*

Then click ‘Submit All Changes’ button at the bottom.

Wait until changes are saved.



Step 5.

Go to ‘Voice’ tab, then go ‘SIP’ tab and change the following settings:

Under ‘RTP Parameters’ category:

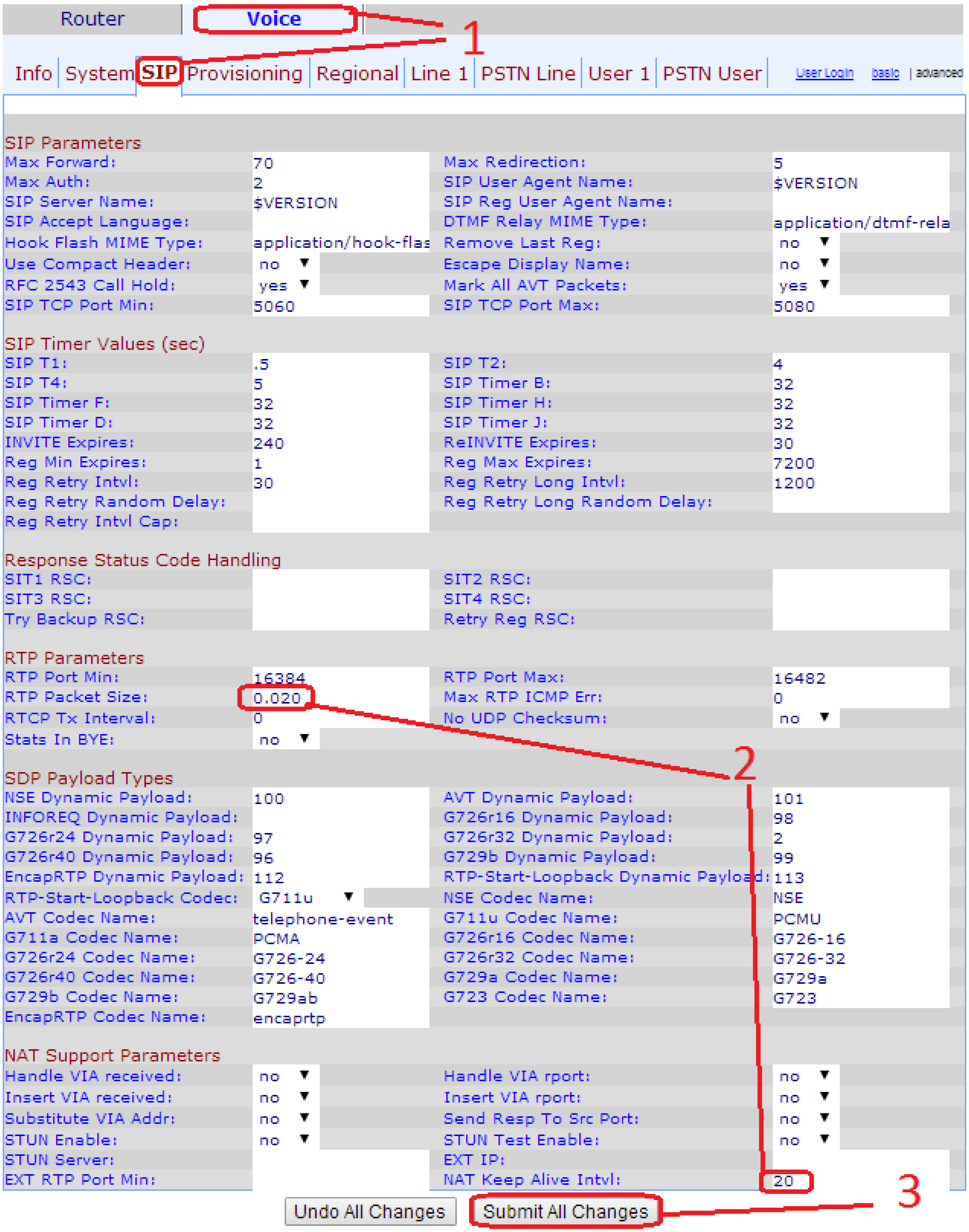
RTP Packet Size: *0.020*

Under ‘NAT Support Parameters’ category:

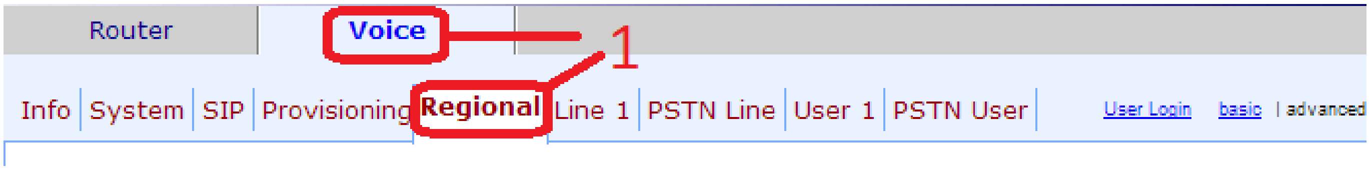
NAT Keep Alive Intvl: *20*

Then click ‘Submit All Changes’ button at the bottom.

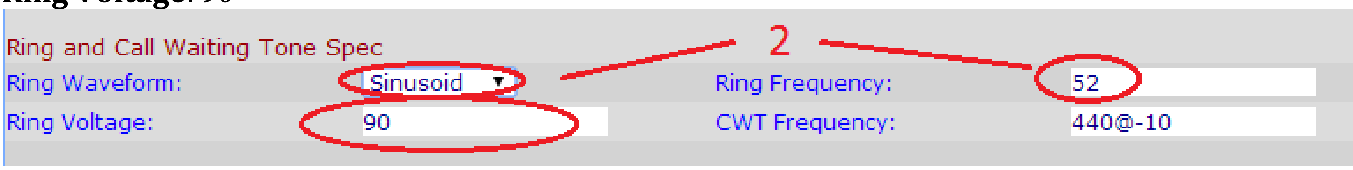
Wait until changes are saved.


Router
Voice
1
Info
System
SIP
Provisioning
Regional
Line 1
PSTN Line
User 1
PSTN User
User Login
basic
advanced
SIP Parameters
Max Forward: 70
Max Redirection: 5
Max Auth: 2
SIP User Agent Name: $VERSION
SIP Server Name: $VERSION
SIP Reg User Agent Name:
SIP Accept Language:
DTMF Relay MIME Type: application/dtmf-rela
Hook Flash MIME Type: application/hook-flas
Remove Last Reg: no
Use Compact Header: no
Escape Display Name: no
RFC 2543 Call Hold: yes
Mark All AVT Packets: yes
SIP TCP Port Min: 5060
SIP TCP Port Max: 5080
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
Reg Retry Random Delay:
Reg Retry Long Random Delay:
Reg Retry Intvl Cap:
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
No UDP Checksum: no
Stats In BYE: no
2
SDP Payload Types
NSE Dynamic Payload: 100
AVT Dynamic Payload: 101
INFOREQ Dynamic Payload:
G726r16 Dynamic Payload: 98
G726r24 Dynamic Payload: 97
G726r32 Dynamic Payload: 2
G726r40 Dynamic Payload: 96
G729b Dynamic Payload: 99
EncapRTP Dynamic Payload: 112
RTP-Start-Loopback Dynamic Payload: 113
RTP-Start-Loopback Codec: G711u
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
G723 Codec Name: G723
EncapRTP Codec Name: encaprtp
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
Undo All Changes
Submit All Changes
3

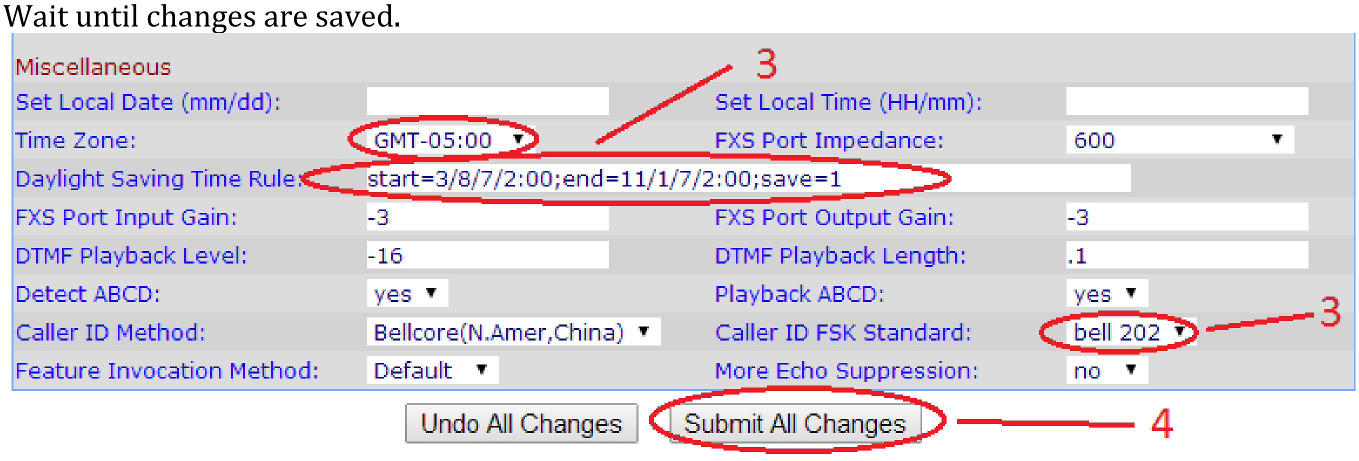
Step 6.
Go to 'Voice' tab, then go 'Regional' tab and change the following settings:



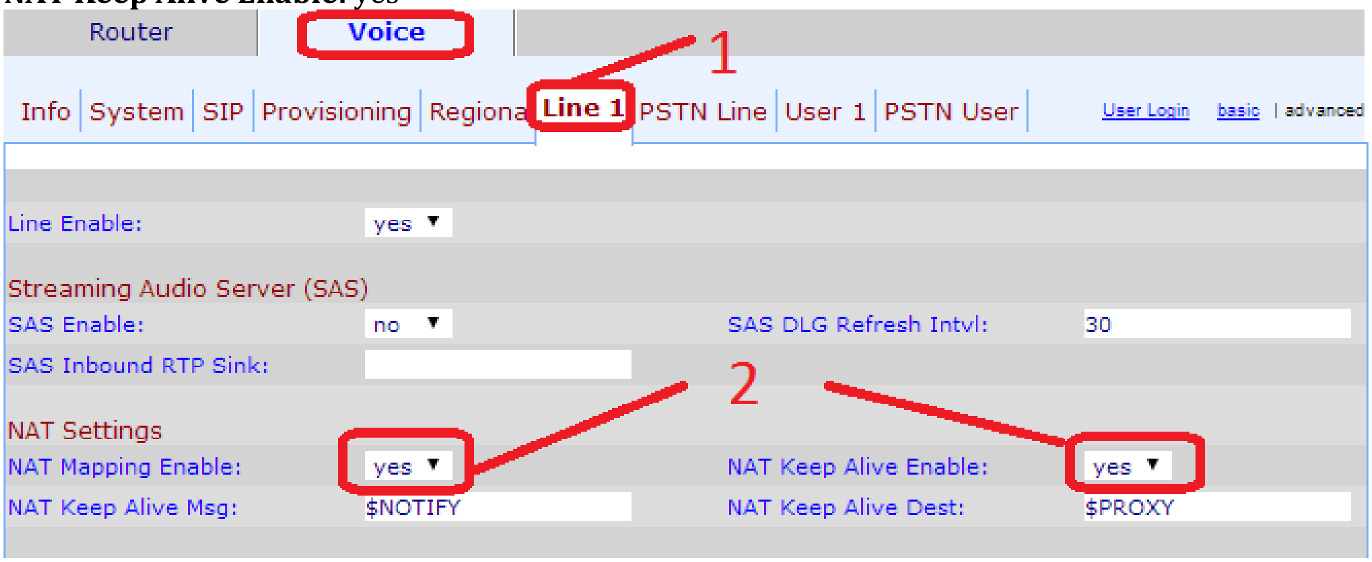
Under 'Ring and Call Waiting Tone Spec' category:
Ring Waveform: *Sinusoid*
Ring Frequency: *52*
Ring Voltage: *90*



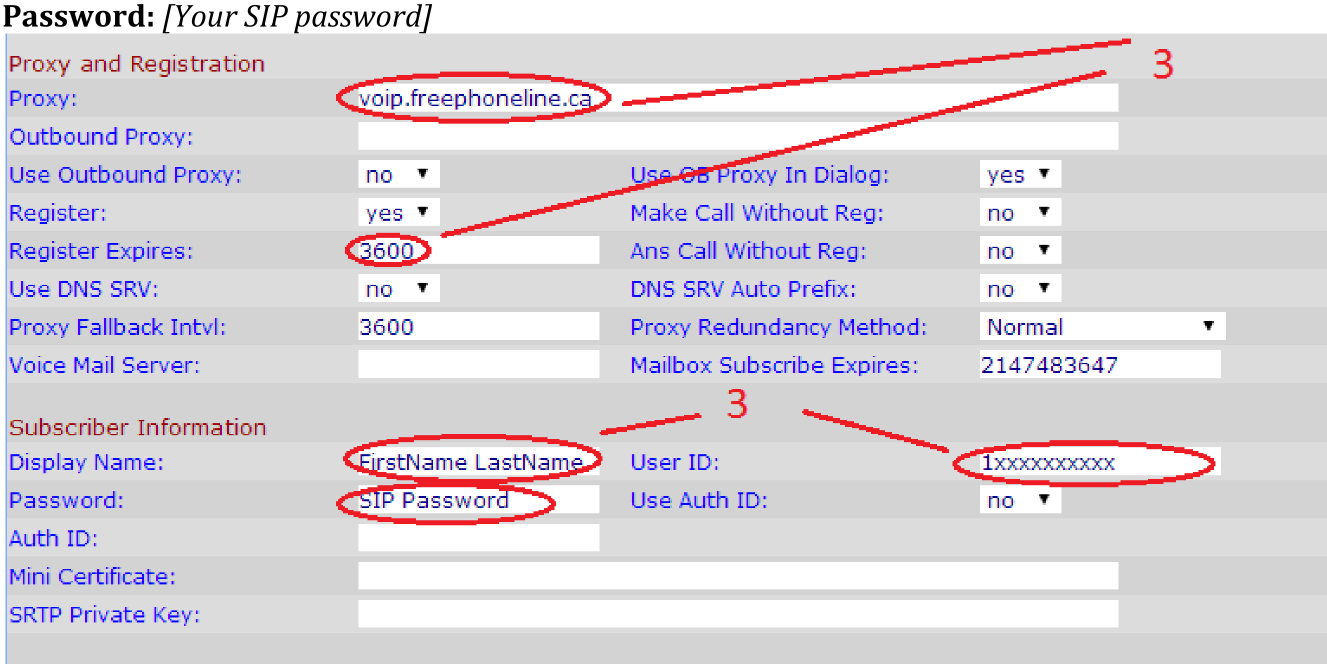
Under 'Miscellaneous' category:
Time Zone: *[Choose appropriate time zone depending where you are located]*
Daylight Savings Time Rule: *start=3/8/7/2:00;end=11/1/7/2:00;save=1*
Then click 'Submit All Changes' button at the bottom.
Wait until changes are saved.



<u>Step 7.</u>
Go to 'Voice' tab, then go '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 <u>ROGERS</u> 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 'Audio Configuration' category:
Preferred Codec: *G711u*
Second Preferred Codec: *G729a*
Third Preferred Codec: *G711u*
Under 'Dial Plan' category:
Dial Plan: *(911|[2-9]xxxxxxxxx|1xxxxxxxxxx|011xxxxxxxxxxxx.|98*|[6-7]x*xxxxxxxxxxx.)*

Then click 'Submit All Changes' button at the bottom.
Wait until changes are saved.

Enjoy your free phone line! ☺