

## **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#'

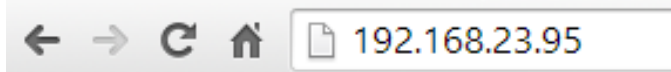
Write down your IP address.

#### **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

Status    Wan Setup    Admin Login    basic    advanced

**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.

The screenshot shows the 'Wan Setup' configuration page. The 'Router' tab is selected at the top, and the 'Wan Setup' sub-tab is active. A red circle and arrow labeled '1' point to the 'Wan Setup' tab. The 'Internet Connection Settings' section is expanded, showing 'Connection Type' set to 'DHCP'. The 'Static IP Settings' section is collapsed. The 'PPPoE Settings' section is collapsed. The 'Optional Settings' section is expanded, showing 'Host Name', 'Primary DNS', 'DNS Server Order' (set to 'Manual'), 'Domain', 'Secondary DNS', 'DNS Query Mode' (set to 'Parallel'), 'Primary NTP Server' (set to 'time.nist.gov'), and 'Secondary NTP Server' (set to 'time.windows.com'). A red circle and arrow labeled '2' point to the 'Optional Settings' section. The 'MAC Clone Settings' section is collapsed. The 'Remote Management' section is expanded, showing 'Enable WAN Web Server' (set to 'yes') and 'WAN Web Server Port' (set to '80'). The 'QoS Settings' section is collapsed. The 'VLAN Settings' section is collapsed. At the bottom, there are two buttons: 'Undo All Changes' and 'Submit All Changes'. A red circle and arrow labeled '3' point to the 'Submit All Changes' button.

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*

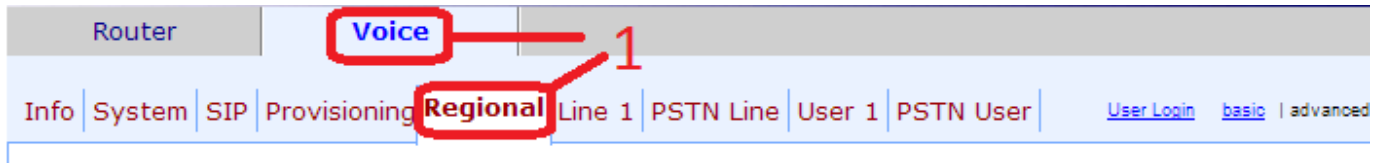
Then click 'Submit All Changes' button at the bottom.

Wait until changes are saved.

Router		Voice	
<a href="#">Info</a>   <a href="#">System</a>   <a href="#">SIP</a>   <a href="#">Provisioning</a>   <a href="#">Regional</a>   <a href="#">Line 1</a>   <a href="#">PSTN Line</a>   <a href="#">User 1</a>   <a href="#">PSTN User</a>   <a href="#">User Login</a>   <a href="#">basic</a>   <a href="#">advanced</a>			
<b>SIP Parameters</b>			
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
<b>SIP Timer Values (sec)</b>			
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:			
<b>Response Status Code Handling</b>			
SIT1 RSC:		SIT2 RSC:	
SIT3 RSC:		SIT4 RSC:	
Try Backup RSC:		Retry Reg RSC:	
<b>RTP Parameters</b>			
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 ▼		
<b>SDP Payload Types</b>			
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:	encaprtsp		
<b>NAT Support Parameters</b>			
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
<a href="#">Undo All Changes</a>		<a href="#">Submit All Changes</a>	

Step 6.

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

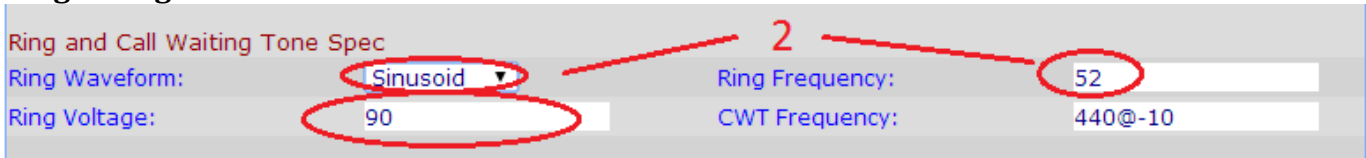


Under 'Ring and Call Waiting Tone Spec' category:

**Ring Waveform:** *Sinusoid*

**Ring Frequency:** 52

**Ring Voltage:** 90



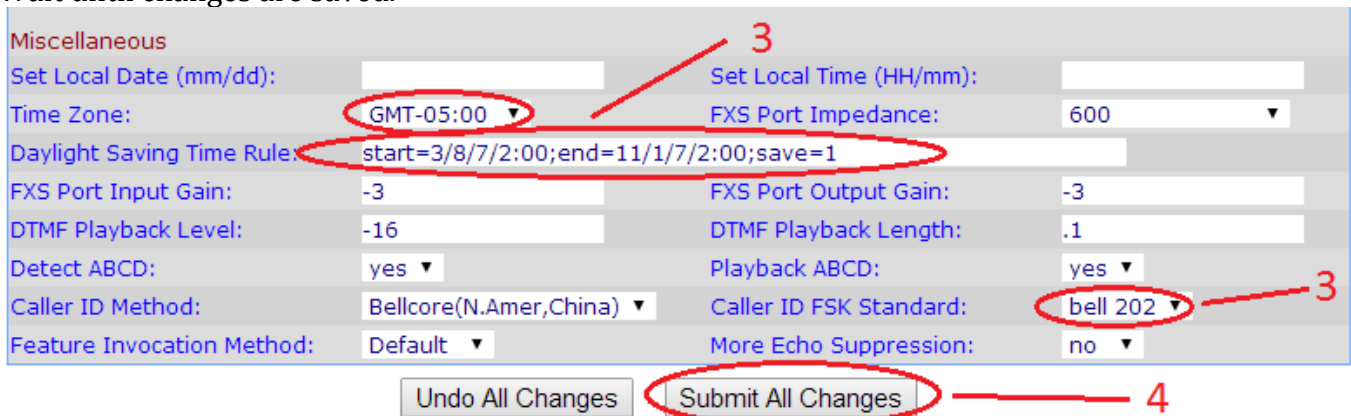
Under 'Miscellaneous' category:

**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.



### Step 7.

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

Router | **Voice**

Info | System | SIP | Provisioning | Regional | **Line 1** | PSTN Line | User 1 | PSTN User | [User Login](#) | [basic](#) | [advanced](#)

Line Enable: yes ▼

Streaming Audio Server (SAS)

SAS Enable: no ▼ SAS DLG Refresh Intvl: 30

SAS Inbound RTP Sink:

NAT Settings

NAT Mapping Enable: yes ▼ NAT Keep Alive Enable: yes ▼

NAT Keep Alive Msg: \$NOTIFY NAT Keep Alive Dest: \$PROXY

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 1xxxxxxxxx]

**Password:** [Your SIP password]

Proxy and Registration

Proxy: voip.freephoneline.ca

Outbound Proxy:

Use Outbound Proxy: no ▼ Use Out Proxy In Dialog: yes ▼

Register: yes ▼ Make Call Without Reg: no ▼

Register Expires: 3600 Ans Call Without Reg: no ▼

Use DNS SRV: no ▼ DNS SRV Auto Prefix: no ▼

Proxy Fallback Intvl: 3600 Proxy Redundancy Method: Normal ▼

Voice Mail Server: Mailbox Subscribe Expires: 2147483647

Subscriber Information

Display Name: FirstName LastName User ID: 1xxxxxxxxx

Password: SIP Password Use Auth ID: no ▼

Auth ID:

Mini Certificate:

SRTP Private Key:

Under 'Audio Configuration' category:

**Preferred Codec:** *G711u*

**Second Preferred Codec:** *G729a*

**Third Preferred Codec:** *G711u*

Under 'Dial Plan' category:

**Dial Plan:** *(911|/[2-9]xxxxxxxx/1xxxxxxxx/011xxxxxxxxxx./98\*|/[6-7]x\*xxxxxxxx.)*

**Then click 'Submit All Changes' button at the bottom.**

Wait until changes are saved.

**Audio Configuration**

Preferred Codec:	G711u ▼	Second Preferred Codec:	G729a ▼
Third Preferred Codec:	G711u ▼	Use Pref Codec Only:	no ▼
Silence Supp Enable:	no ▼	Silence Threshold:	medium ▼
G729a Enable:	yes ▼	Echo Canc Enable:	yes ▼
G723 Enable:	yes ▼	Echo Canc Adapt Enable:	yes ▼
G726-16 Enable:	yes ▼	Echo Supp Enable:	yes ▼
G726-24 Enable:	yes ▼	FAX CED Detect Enable:	yes ▼
G726-32 Enable:	yes ▼	FAX CNG Detect Enable:	yes ▼
G726-40 Enable:	yes ▼	FAX Passthru Codec:	G711u ▼
DTMF Process INFO:	yes ▼	FAX Codec Symmetric:	yes ▼
DTMF Process AVT:	yes ▼	FAX Passthru Method:	NSE ▼
DTMF Tx Method:	Auto ▼	DTMF Tx Mode:	Strict ▼
DTMF Tx Strict Hold Off Time:	40	FAX Process NSE:	yes ▼
Hook Flash Tx Method:	None ▼	FAX Disable ECAN:	no ▼
Release Unused Codec:	yes ▼	FAX Enable T38:	yes ▼
FAX T38 Redundancy:	1 ▼	FAX T38 V29 Only:	no ▼
FAX Tone Detect Mode:	caller or callee ▼	FAX Enable T38 Extend CED:	no ▼
Audio Dump Option1:	none ▼	Audio Dump Option2:	none ▼

**Dial Plan**

Dial Plan: (911|/[2-9]xxxxxxxx/1xxxxxxxx/011xxxxxxxxxx./98\*|/[6-7]x\*xxxxxxxx.)

Enable IP Dialing: no ▼      Emergency Number:

**FXS Port Polarity Configuration**

Idle Polarity: Forward ▼      Caller Conn Polarity: Forward ▼

Callee Conn Polarity: Forward ▼

Undo All Changes      Submit All Changes

Enjoy your free phone line! ☺