

Linksys SPA2102 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 SPA2102 device.

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

Step 2.

By default settings Linksys SPA2102 WAN access is disabled.

You have to enable this setting before you access web-based utility.

From the phone attached to your Linksys SPA2102 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#'

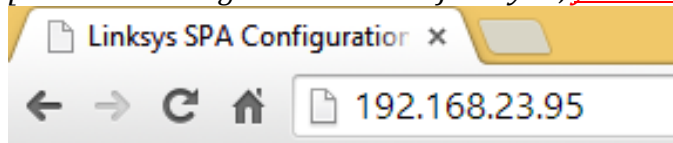
Write down your IP address.

Step 3.

On the PC/Mac connected to the same network as your Linksys SPA2102 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'.

Router Voice 2

Status Wan Setup 1 [Admin Login](#) basic [advanced](#)

Product Information

Product Name:	SPA-2102	Serial Number:
Software Version:	3.3.6	Hardware Version:	1.3.5(a)
MAC Address:	Client Certificate:	Installed
Customization:	Open		

System Status

Current Time:	1/1/2003 12:33:48	Elapsed Time:	00:22:54
Wan Connection Type:	DHCP	Current IP:	192.168.23.95
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:	0
Broadcast Bytes Sent:	0	Broadcast Pkts Recv:	8388
Broadcast Bytes Recv:	1113302	Broadcast Pkts Dropped:	0
Broadcast Bytes Dropped:	0		

[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 for a router. The 'Router' and 'Wan Setup' tabs are circled in red and labeled with a red '1'. The 'Internet Connection Settings' section is labeled with a red '2'. Within this section, the 'Primary NTP Server' is set to 'time.nist.gov' and the 'Secondary NTP Server' is set to 'time.windows.com', both circled in red. The 'Submit All Changes' button at the bottom is circled in red and labeled with a red '3'.

Section	Field	Value
Internet Connection Settings	Connection Type:	DHCP
	Static IP:	
	Gateway:	
	NetMask:	
PPPoE Settings	PPPOE Login Name:	
	PPPOE Login Password:	
Optional Settings	Host Name:	
	Primary DNS:	
	DNS Server Order:	Manual
	Primary NTP Server:	time.nist.gov
MAC Clone Settings	Enable MAC Clone Service:	no
	Cloned MAC Address:	
Remote Management	Enable WAN Web Server:	yes
	WAN Web Server Port:	80
QOS Settings	QOS QDisc:	NONE
	Maximum Uplink Speed:	128 (Kbps)
VLAN Settings	Enable VLAN:	no
	VLAN ID:	1 [0x000-0xFFFF]

Buttons: Undo All Changes, Submit All Changes

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 Line 2 User 1 User 2 [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 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	No UDP Checksum:	no ▼
Stats In BYE:	no ▼		

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
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		

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: *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.

Router | **Voice** | 1

Info | System | SIP | Provisioning | **Regional** | Line 1 | Line 2 | User 1 | User 2 | [User Login](#) | [basic](#) | [advanced](#)

Ring and Call Waiting Tone Spec

Ring Waveform: **Sinusoid** | Ring Frequency: **52**

Ring Voltage: **90** | CWT Frequency: 440@-10

Miscellaneous

Set Local Date (mm/dd): | Set Local Time (HH/mm): |

Time Zone: **GMT-05:00** | FXS Port Impedance: 600

Daylight Saving Time Rule: **start=3/8/7/2:00;end=11/1/7/2:00;save=1**

FXS Port Input Gain: -3 | FXS Port Output Gain: -3

DTMF Playback Level: -16 | DTMF Playback Length: .1

Detect ABCD: yes | Playback ABCD: yes

Caller ID Method: Bellcore(N.Amer,China) | Caller ID FSK Standard: **bell 202**

Feature Invocation Method: Default | More Echo Suppression: no

[Undo All Changes](#) | [Submit All Changes](#) 4

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** | 1

Info | System | SIP | Provisioning | Regional | **Line 1** | Line 2 | User 1 | User 2 | [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

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]*

The screenshot shows a configuration interface with two main sections: 'Proxy and Registration' and 'Subscriber Information'. In the 'Proxy and Registration' section, the 'Proxy' field is set to 'voip.freephoneline.ca' and 'Register Expires' is set to '3600'. In the 'Subscriber Information' section, the 'Display Name' is 'FirstName LastName', the 'Password' is 'SIP Password', and the 'User ID' is '1xxxxxxxxx'. Red circles highlight these five fields. Red arrows originate from a red number '3' and point to each of the five highlighted fields.

Proxy and Registration	
Proxy:	voip.freephoneline.ca
Outbound Proxy:	
Use Outbound Proxy:	no
Register:	yes
Register Expires:	3600
Use DNS SRV:	no
Proxy Fallback Intvl:	3600
Voice Mail Server:	
Use GB Proxy In Dialog:	yes
Make Call Without Reg:	no
Ans Call Without Reg:	no
DNS SRV Auto Prefix:	no
Proxy Redundancy Method:	Normal
Mailbox Subscribe Expires:	2147483647

Subscriber Information	
Display Name:	FirstName LastName
Password:	SIP Password
Auth ID:	
Mini Certificate:	
SRTP Private Key:	
User ID:	1xxxxxxxxx
Use Auth ID:	no

Under 'Dial Plan' category:

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

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

Wait until changes are saved.

Audio Configuration

Preferred Codec:	G711u	Silence Supp Enable:	no
Use Pref Codec Only:	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	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 Tone Detect Mode:	caller or callee

Dial Plan

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

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

Step 8.

Go to 'Voice' tab, then go to 'Line 2' tab:

Line Enable: no

Router Voice

Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2 | [User Login](#) | [basic](#) | [advanced](#)

Line Enable: no

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

Wait until changes are saved.

Enjoy your free phone line! ☺