

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#'

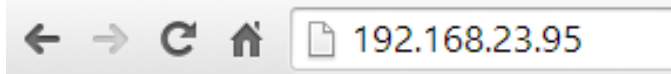
Write down your IP address.

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'.

LINKSYS®
A Division of Cisco Systems, Inc.

Firmware Version: 2.0.14(LSVa)

Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info System User 1 User 2

Basic View (switch to advanced view) Admin Login

System Information

DHCP:	Enabled	Current IP:	192.168.1.100
Host Name:	LinksysPAP	Domain:	
Current Netmask:	255.255.255.0	Current Gateway:	192.168.1.1
Primary DNS:	192.168.1.1		
Secondary DNS:			

Product Information

Product Name:	PAP2-NA	Serial Number:	90 C8 9
Software Version:	2.0.14(LSVa)	Hardware Version:	0.03.4
MAC Address:	121 C3 D	Client Certificate:	Installed
Customization:	Not Customized		

System Status

Current Time:	1/1/2003 12:02:54	Elapsed Time:	00:02:54
Broadcast Pkts Sent:	4	Broadcast Bytes Sent:	1368
Broadcast Pkts Recv:	82	Broadcast Bytes Recv:	9530
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	0	RTP Bytes Sent:	0
RTP Packets Recv:	0	RTP Bytes Recv:	0
SIP Messages Sent:	0	SIP Bytes Sent:	0
SIP Messages Recv:	0	SIP Bytes Recv:	0
External IP:			

Line 1 Status

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.

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

System Configuration

Restricted Access Domains:

Enable Web Server: Web Server Port:

Enable Web Admin Access: Admin Passwd:

User Password:

Internet Connection Type

DHCP: Static IP: NetMask:

Gateway:

Optional Network Configuration

HostName: Domain:

Primary DNS: Secondary DNS:

DNS Server Order: DNS Query Mode:

Syslog Server: Debug Server:

Debug Level: Primary NTP Server:

Secondary NTP Server:

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Cisco Systems

Step 5.

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](#)

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		

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
G729o 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
G729o 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

Step 6.

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*

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Voice

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

Advanced View (switch to basic view) User Login

Ring and Call Waiting Tone Spec

Ring Waveform: Sinusoid

Ring Frequency: 52

Ring Voltage: 90

CWT Frequency: 440@-10

Synchronized Ring: no

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.

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)

FXS Port Power Limit: 3

Save Settings Cancel Settings

Step 7.

Go to 'Line 1' tab and change the following settings:

Under 'NAT Settings' category:

NAT Mapping Enable: *yes*

NAT Keep Alive Enable: *yes*

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

Streaming Audio Server (SAS)

NAT Settings

Line Enable: yes ▼

SAS Enable: no ▼ SAS DLG Refresh Intvl: 30

SAS Inbound RTP Sink:

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

Outbound Proxy:

Register: yes ▼

Register Expires: 3600

Use DNS SRV: no ▼

Proxy Fallback Intvl: 3600

Use Outbound Proxy: no ▼

Use OB Proxy In Dialog: yes ▼

Make Call Without Reg: no ▼

Ans Call Without Reg: no ▼

DNS SRV Auto Prefix: no ▼

Voice Mail Server:

Subscriber Information

Display Name: Firstname Lastname

Password: Your SIP pw

Auth ID:

User ID: 1xxxxxxxxx

Use Auth ID: no ▼

Under 'Dial Plan' category:

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

Then click 'Save Settings' button at the bottom.

Wait until changes are saved.

Audio Configuration

Preferred Codec: <input type="text" value="G711u"/>	Silence Supp Enable: <input type="text" value="no"/>
Use Pref Codec Only: <input type="text" value="no"/>	Silence Threshold: <input type="text" value="medium"/>
G729a Enable: <input type="text" value="yes"/>	Echo Canc Enable: <input type="text" value="yes"/>
Echo Canc Adapt Enable: <input type="text" value="yes"/>	G726-16 Enable: <input type="text" value="yes"/>
Echo Supp Enable: <input type="text" value="yes"/>	G726-24 Enable: <input type="text" value="yes"/>
FAX CED Detect Enable: <input type="text" value="yes"/>	G726-32 Enable: <input type="text" value="yes"/>
FAX CNG Detect Enable: <input type="text" value="yes"/>	G726-40 Enable: <input type="text" value="yes"/>
FAX Passthru Codec: <input type="text" value="G711u"/>	FAX Codec Symmetric: <input type="text" value="yes"/>
FAX Passthru Method: <input type="text" value="NSE"/>	DTMF Tx Method: <input type="text" value="Auto"/>
FAX Process NSE: <input type="text" value="yes"/>	Hook Flash Tx Method: <input type="text" value="None"/>
Release Unused Codec: <input type="text" value="yes"/>	

Dial Plan

Dial Plan:

Enable IP Dialing:

FXS Port Polarity Configuration

Idle Polarity:

Caller Conn Polarity:

Callee Conn Polarity:

Save Settings **Cancel Settings**

Step 8.

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

Line Enable: no

Voice

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Advanced View (switch to basic view) [User Login](#)

Line Enable:

Save Settings **Cancel Settings**

Then click 'Save Settings' button at the bottom.

Wait until changes are saved.

Enjoy your free phone line! ☺