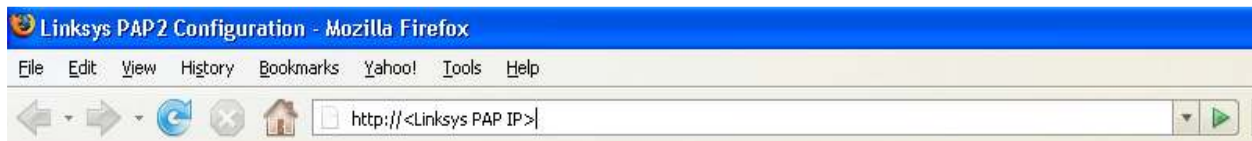


PAP2T/PAP2 CONFIGURATION



The Linksys Internet Phone Adapter enables high-quality feature-rich VoIP (voice over IP) service through your broadband Internet connection. Just plug it into your home Router or Gateway and use the two standard telephone ports to connect analog phones or use one of the ports for a fax machine. Each phone port operates independently, with separate phone service and phone numbers, like having two telephone lines. You'll get clear reception and a reliable fax connection, even while using the Internet at the same time.

1. Connect the analogical telephone line to the adapter plugging in the telephone cable in the appropriate port (Phone 1)
2. Connect the adapter to the network plugging in the patch cable in the appropriate port (Ethernet)
3. Dial **** (4 asterisks), then, after the electronic voice's request, dial 110#
4. The electronic voice will give you the IP address assigned through the DHCP spelling each single digit and every dot between the digits (for example, 192.168.1.101 will be one-nine-two-dot-one-six-eight-dot-one-dot-one-zero-one)
5. Key in the IP address in your web browser: to open the adapter's (PAP2T) web interface



- To configure Silverback ASP account extension, select Admin Login, on the right high corner, then select Line 1



Proxy and Registration

In the field Proxy input: sip.silverbackasp.net

Subscriber Information

Displayname: Key in your company/owners name
 User ID: key in your Silverback ASP extension (e.g. 2001)
 Password: key in your Silverback ASP extension password
 Auth ID: Key in your Silverback ASP extension (same with your User ID)
 Use Auth ID: yes

Audio Configuration

Preferred Codec: choose G723

LINKSYS
A Division of Cisco Systems, Inc. Firmware Version: 1.1.30.31

Phone Adapter with 2 Ports for Voice-Over-IP PAP2

Voice

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

Basic View (switch to advanced view) User Login

SIP Settings

Line Enable: yes no

SIP Port:

Proxy:

Register: yes no

Register Expires:

Make Call Without Reg: no yes

Ans Call Without Reg: no yes

Subscriber Information

Display Name:

User ID:

Password:

Use Auth ID: no yes

Auth ID:

Supplementary Service Subscription

Call Waiting Serv: yes no

Block CID Serv: yes no

Audio Configuration

MMI Serv: yes no

VMM Serv: yes no

Preferred Codec:

Silence Supp Enable: no yes

Use Pref Codec Only: no yes

FAX CED Detect Enable: yes no

DTMF Tx Method:

Save Settings Cancel Settings

Cisco Settings

7. Configure "Line 2" as well with the same configuration. But use unique configuration for **UserID/Password** and **SIP Port** in Line1 and Line2.
8. Configure unique accounts for UserID in Line1 and Line2.
9. SIP Port should be something different than Line 1,
10. Line 1 SIP Port Default: 5060 Line 2 SIP Port Default: 5061

11. Then go to the “SIP” tab for the STUN configuration. At the NAT Support Parameters key in “stun.silverbackasp.net:3478”

NAT Support Parameters	
Handle VIA received:	<input type="text" value="no"/>
Insert VIA received:	<input type="text" value="no"/>
Substitute VIA Addr:	<input type="text" value="no"/>
STUN Enable:	<input type="text" value="yes"/>
STUN Server:	<input type="text" value="stun.silverbackasp.net:3478"/>
EXT RTP Port Min:	<input type="text"/>
Handle VIA rport:	<input type="text" value="no"/>
Insert VIA rport:	<input type="text" value="no"/>
Send Resp To Src Port:	<input type="text" value="no"/>
STUN Test Enable:	<input type="text" value="yes"/>
EXT IP:	<input type="text"/>
NAT Keep Alive Intvl:	<input type="text" value="15"/>

12. Now press “Save Settings” button. Device would restart and you’ll be ready to call