

# Sipura SPA-841 FXS Configuration

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1. Open your web browser and navigate to the URL;

**http://<Linksys PAP IP>/admin/advanced**

2. Go to "SIP" Page and check the configuration to match the below highlighted area;

The screenshot shows the SIP configuration page with the following parameters:

SIP Parameters			
Max Forward:	70	Max Redirection:	5
Max Auth:	2	SIP User Agent Name:	\$VERSION

Three arrows point from the SIP Parameters section to the NAT Support Parameters section.

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:	15

3. Go to "LINE 1", and configure it to look as displayed below:

General

Line Enable:

Share Line Appearance

Share Ext:  Shared User ID:   
Subscription Expires:

NAT Settings

NAT Mapping Enable:  NAT Keep Alive Enable:   
NAT Keep Alive Msg:  NAT Keep Alive Dest:

Network Settings

SIP TOS/DiffServ Value:  SIP CoS Value:   
RTP TOS/DiffServ Value:  RTP CoS Value:   
Network Jitter Level:

SIP Settings

SIP Port:  SIP 100REL Enable:   
EXT SIP Port:  Auth Resync-Reboot:   
SIP Proxy-Require:  SIP Remote-Party-ID:   
Referor Bye Delay:  Refer-To Target Contact:   
Referee Bye Delay:  SIP Debug Option:   
Refer Target Bye Delay:  Sticky 183:

Call Feature Settings

Blind Attn-Xfer Enable:  MOH Server:   
Message Waiting:  Auth Page:   
Default Ring:  Auth Page Realm:   
Conference Bridge URL:  Auth Page Password:   
Voice Mail Server:  State Agent:   
CFWD Notify Serv:  CFWD Notifier:

Proxy and Registration

Proxy:  Use Outbound Proxy:   
Outbound Proxy:  Use OB Proxy In Dialog:   
Register:  Make Call Without Reg:   
Register Expires:  Ans Call Without Reg:   
Use DNS SRV:  DNS SRV Auto Prefix:

**Ref. A**



Proxy Fallback Intvl:  Proxy Redundancy Method:

Subscriber Information

Display Name:  User ID:   
Password:  Use Auth ID:   
Auth ID:   
Mini Certificate:   
SRTP Private Key:

Audio Configuration

Preferred Codec:  Use Pref Codec Only:   
Use Pref Codec Only:  G723 Enable:   
G726-16 Enable:  G726-24 Enable:

Ref A:

Proxy: [sip.yapcalls.com](http://sip.yapcalls.com)

Ref B:

Preferred codecs are **G723** and **G729a**

Use preferred codec only: **YES**

Note: If you're using PAP2T, then you're able to set both lines on G729a. However incase of PAP2, this cannot be done due to a limitation of the device itself.

Ref C:

Dial Plan:

(\*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxxx.)

4. Configure "Line 2" as well to with same configuration. The Only difference would be SIP PORT: should be something different than line 1,

Line 1 SIP Port Default: 5060

Line 2 SIP Port Default: 5061

5. Now press "Save Settings" button. Device would restart and you'll be ready to call!

If you experience any problem with configurations, please do feel free to call for support, and we'll be very glad to assist you.