

Linksys Sipura FXS Gateways

Linksys SPA3102

The following configuration has not been tested for PSTN failover. The FXS port is registering to a sipXcom 14.10 PBX and handles inbound/outbound fax transmissions as well as voice.

The first page as seen below shows the firmware and hardware version that this unit was running.

Info System SIP Provisioning Regional Line 1 PSTN Line User 1 PSTN User			
Product Information			
Product Name:	SPA-3102	Serial Number:	FM60P00995
Software Version:	3.3.6(070)	Hardware Version:	1.1.3
MAC Address:	00000003FCB	Clear Certificates:	Installed
Customization:	Open		
System Status			
Current Time:	4/1/2010 21:05:53	Elapsed Time:	1 day and 10:06:35
RTP Packets Sent:	6639	RTP Bytes Sent:	1361940
RTP Packets Recv:	8636	RTP Bytes Recv:	1381740
SIP Messages Sent:	126	SIP Bytes Sent:	59457
SIP Messages Recv:	126	SIP Bytes Recv:	77557
Line 1 Status			
Hook State:	On	Registration State:	Registered
Last Registration At:	4/1/2010 21:19:02	Next Registration In:	1279 s
Message Strategy:	No	Call Back Active:	No
Last Called Number:		Last Caller Number:	
PSTN Line Status			
Hook State:	On	Line Voltage:	0 (V)
Loop Current:	0.0 (mA)	Registration State:	Not Registered
Last Registration At:		Next Registration In:	
Last Called VSP Number:		Last Called PTTN Number:	
Last VSP Caller:		Last PTTN Caller:	
Last PSTN Disconnect Reason:		PSTN Activity Timer:	30000 (ms)
Registered SIP Trunk:		Call Type:	

Click on **Admin Login** and **Advanced** and then click on the **SIP** tab under the **Voice** settings. The following page will be displayed:

LINKSYS® A Division of Cisco Systems, Inc. Linksys Phone Adapter Configuration			
Router Voice			
Info System SIP Provisioning Regional Line 1 PSTN Line User 1 PSTN User			
SIP Parameters			
Max Proxies:	70	Max Redirects:	5
Max Auth:	2	SIP User Agent Name:	SIPVERSION
SIP Server Name:	SIPVERSION	SIP Reg User Agent Name:	SIPVERSION
SIP Server Language:	application/javascript	DTMF Relay Mode Type:	application/dtmfrelay
Hook Flash Hold Type:	application/hook-flash	Remove Last Reg:	no
Use Compact Headers:	no	Escape Outgoing Names:	no
RFC 2543 Call Hold:	yes	Mark All AVT Packets:	yes
SIP Timer Values (sec)			
SIP T1:	5	SIP T2:	4
SIP T3:	4	SIP Timer B1:	32
SIP Timer F1:	32	SIP Timer F2:	32
SIP Timer R1:	32	SIP Timer R2:	32
INVITE Expires:	240	Reinvite Expires:	30
Reg In Expires:	1	Reg In Expires:	7200
Reg Party Inact:	30	Reg Party Long Inact:	1200
Response Status Code Handling			
SIP183C:		SIP183C:	
SIP184C:		SIP184C:	
Try Before RSC:		Retry Reg RSC:	
RTP Parameters			
RTP Port Min:	16384	RTP Port Max:	16462
RTP Packet Size:	0.020	Max RTP Send Buf:	0
RTP Tx Interval:	0	No UDP Checksum:	no
Send In BIC:	no		
SIP Preferred Types			
183 (Dynamic Payload):	100	AVT Dynamic Payload:	101
183 (Static Payload):	100	DTMF Dynamic Payload:	99
183 (QoS Dynamic Payload):	97	173 (QoS Dynamic Payload):	2
183 (QoS Static Payload):	96	173 (QoS Dynamic Payload):	99
183 Code Name:	183	AVT Code Name:	Telephone-event
173 (QoS Code Name):	173	173 (QoS Code Name):	173
173 (QoS Code Name):	173-16	173 (QoS Code Name):	173-24
173 (QoS Code Name):	173-32	173 (QoS Code Name):	173-40
173 (QoS Code Name):	173-48	173 (QoS Code Name):	173-56
173 (QoS Code Name):	173-64	173 (QoS Code Name):	173-72
173 (QoS Code Name):	173-80	173 (QoS Code Name):	173-88
173 (QoS Code Name):	173-96	173 (QoS Code Name):	173-104
NAT Support Parameters			
Handle VSP received:	no	Handle VSP (port):	no
Send VSP received:	no	Send VSP (port):	no
Substitute VSP Addr:	no	Send Pcap To Src Port:	no
STUN Enable:	no	STUN Use Enable:	no
STUN Server:		STUN:	
STUN Port:		STUN Keep Alive Interval:	15

Set RTP Packet Size to 0.020 and then click on the **Submit All Changes** button at the bottom of the page.

Next, click on the **Line 1** tab. Set the **Line Enable** to **yes**, set any QoS settings you may require, set the **SIP Transport** to **UDP**, set the **Proxy** to your **SIP domain** (for SRV sipXcom setup), set **Reigster** to **Yes**, set **Use DNS SRV** to **Yes**, set DNS SRV Auto Prefix to **Yes**.

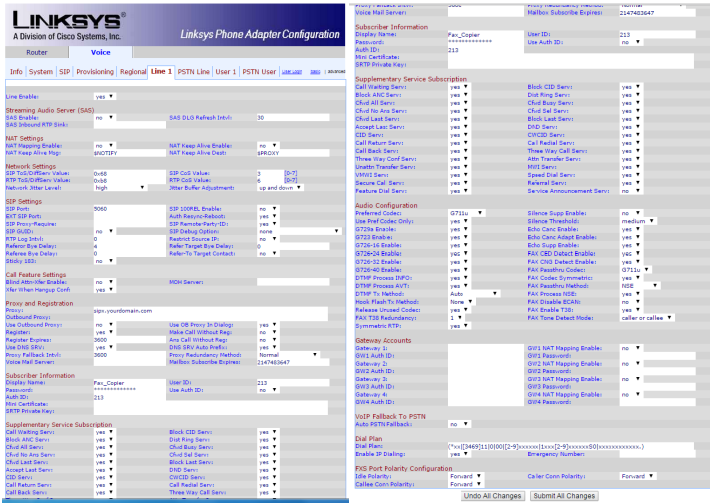
Set the **Display Name** to what you want local phone users to see for a user name, set **User ID** to the user's PBX extension, set the **Password** to the SIP password (not the PIN!), set **Use Auth ID** to **no**, set **Auth ID** to the user's PBX extension.

Set the **Preferred Codec** to **G711u**, set **Use Pref Codec Only** to **yes**.

The following dial plan will dial just about any valid number (depending on you Dial Plan in the PBX) . Just before the end of the page set the **Dial Plan** to: (*xx[[3469]11|0|00][2-9]xxxxxx|1xxx[2-9]xxxxxxS0]xxxxxxxxxxxx.)

Click on 'Submit All Changes' when done with this page.

All of these settings can be seen in the following three screen shots:



If your DNS is configured properly the gateway should register and be able to make calls at this point.

Linksys / Sipura PAP2

We need more detail here...

PAP2 Notes:

PAP2 Settings to adjust.

Under the sip tab:

RTP Packet Size: 0.020

Under the line settings:

DTMF Process INFO: no
 DTMF Process AVT: no
 DTMF Tx Method: Auto
 DTMF Tx Mode: Strict

I think that is all I changed from the defaults, and these settings work on both of my PAP2's. For some reason this does not work consistent with all the Linksys/Sipura models, for example I ended up using different settings on a SPA 2100.