

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 Pcty Inact: | 30 | Reg Pcty Long Inact: | 1200 |
| Response Status Code Handling | | | |
| SIP1 RSC: | | SIP1 RSC: | |
| SIP4 RSC: | | SIP4 RSC: | |
| Tr. Backlog RSC: | | Reg. RSC: | |
| RTP Parameters | | | |
| RTP Port Size: | 16384 | RTP Port Size: | 16482 |
| RTP Packet Size: | 0.020 | Max RTP Size: | 0 |
| RTP Tr. Interval: | 0 | No UDP Checksum: | no |
| Max In-BK: | no | | |
| SIP Preferred Types | | | |
| 802 Dynamic Payload: | 100 | AVT Dynamic Payload: | 101 |
| 80282 Dynamic Payload: | 99 | 80282 Dynamic Payload: | 99 |
| 80284 Dynamic Payload: | 97 | 80282 Dynamic Payload: | 2 |
| 80284 Dynamic Payload: | 96 | 80282 Dynamic Payload: | 99 |
| 802 Code Name: | 8028 | AVT Code Name: | telephone-event |
| 8028 Code Name: | 8028 | 8028 Code Name: | 8028 |
| 8028 Code Name: | 0726-16 | 8028 Code Name: | 0726-24 |
| 8028 Code Name: | 0726-02 | 8028 Code Name: | 0726-40 |
| 8028 Code Name: | 0726 | 8028 Code Name: | 0726b |
| 8028 Code Name: | 0726 | | |
| NAT Support Parameters | | | |
| Handle VSP received: | no | Handle VSP (port): | no |
| Send VSP received: | no | Send VSP (port): | no |
| Substitute VSP Addr: | no | Send Pcty To Src Port: | no |
| STUN Enable: | no | STUN Use Enable: | no |
| STUN Server: | no | STUN: | no |
| STUN Port: | no | STUN Keep Alive Inact: | 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**.

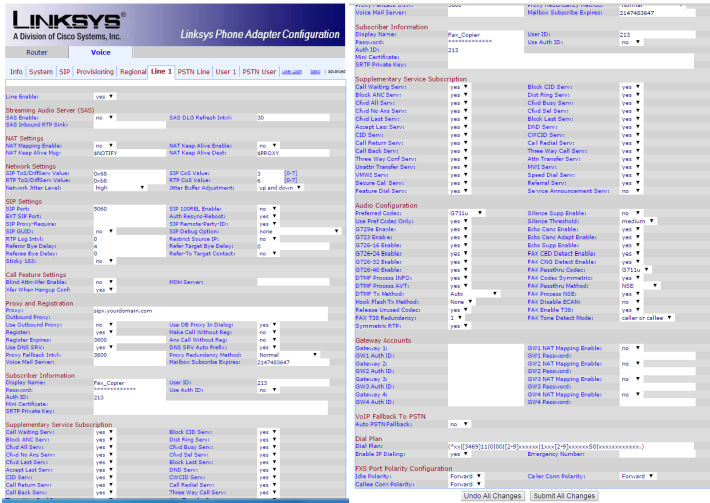
Set the **Display Name** to what you want local phone users to see for a user name, set **User ID** to the user's PBX extension, set the **Password** to the SIP password (not the PIN!), set **Use Auth ID** to **no**, set **Auth ID** to the user's PBX extension.

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.