

Guide to the Linksys SPA-3102



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The Linksys SPA-3102 is a cool little device that can convert your analog phone line into a SIP trunk into your trixbox or Asterisk PBX system. The problem is most people have a ton of problems getting them to work properly.

While there are a ton of settings, there are really only about 7 settings that need to be setup to make this work.

In this article, we will show you the idiots guide to making this cool product work within a trixbox or FreePBX environment. We will walk you through the FreePBX setup of the SIP trunk configuration and then show you the few settings that you must set in order to make the SPA-3102 work. Before starting, go to <http://linksys.com> and get the latest firmware and update your device, then read the article and get your SPA-3102 working within just a few minutes.



Tags: SIP, Asterisk, VoIP, trixbox, Asterisk, Linksys, PBX

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Setup

Go into your FreePBX interface, click on the Setup tab, then go to the Trunk menu. From here, add a new SIP trunk. The settings are pretty simple here but do need some explanation:

General Settings

Outbound Caller ID: "Your Company" <yourphonenumber>

Within quotes, put in your caller ID name information and within the greater-than/less-than brackets put in your outbound caller id phone number. An example of this is as follows:

Outbound Caller ID: "Acme Widgets" <9495551212>

Maximum Channels: 1

You need to set the Maximum channels to 1 since the device can only handle one concurrent call at a time.

Outgoing Settings

Trunk Name: SPA3102

PEER Details:

```
allow=ulaw
canreinvite=no
context=from-pstn
disallow=all
host=10.10.10.50
insecure=very
nat=yes
port=5060
qualify=yes
type=peer
```

General Settings

Outbound Caller ID:

Never Override CallerID:

Maximum channels:

Outgoing Settings

Trunk Name:

PEER Details:

```
allow=ulaw
canreinvite=no
context=from-pstn
disallow=all
host=10.10.10.50
insecure=very
nat=yes
port=5060
qualify=yes
type=peer
```

Incoming Settings

USER Context:

USER Details:

```
allow=gsm|ilbc|ulaw
canreinvite=no
context=from-pstn
disallow=all
host=192.168.0.252
insecure=very
nat=yes
port=5060
type=user
```

Unless you are using G729, you will need to allow ULAW (G711) as that is the default Codec for the SPA-3102. The host address is the IP address of the SPA-3102, if this device is outside of your local network, then set nat=yes, otherwise if it is on the local network, use nat=no.

Incoming Settings

User Context: SPA3102_In

USER Details:

allow=ulaw

canreinvite=no

context=from-pstn

disallow=all

host=192.168.0.252

insecure=very

nat=yes

port=5060

type=user

In the incoming settings, the host address needs to be the IP address of your Asterisk/trixbox PBX system, again use nat=yes if the SPA-3102 is on another subnet or nat=no if the SPA-3102 is on the same subnet as the PBX.

Submit your changes and click the red apply bar, your trixbox/FreePBX system is ready to go, now you need to configure the SPA-3102.

SPA-3102 Configuration

On the rear of the SPA-3102 are four ports, you should plug the phone line into the line port, an analog phone into the phone port, and connect a cable from the Ethernet port to your computer. This will assign your computer an IP address, then you can access the web interface of the SPA-3102 at <http://192.160.0.1>.

If you want the device to sit on your network and not hand out DHCP addresses, plug your regular network into the Internet port, if you have a DHCP server it will assign an IP address to the device or you can set it manually. If you are putting it as a device on your network, be sure and turn on Enable WAN Web Interface otherwise the only way to configure it is to plug a cable from the device to a computer.



Here comes the fun

Now, to configure it as a SIP trunk (this article does explain how to use it as an ATA) you need to click on the Voice tab, and then on the PSTN Line tab, all of the settings are going to changed here.

Proxy and Registration

This section tells the SPA-3102 how to handle calls back and forth, since we are not using authentication, we need to disable those settings. First, set the Proxy setting to point to IP address of your PBX box, then change the settings in the section to match the list I will give you here, you should only have to adjust a few settings, but I will give you all of them to make sure you have everything setuip right:

Use Outbound Proxy: No

Register: No

Register Expires: 3600

Use DNS SRV: No
 Proxy Fallback Intvl: 3600
 Use OB Proxy in Dialog: Yes
 Make Call Without Reg: Yes
 Ans Call Without Reg: Yes
 DNS SRV Auto Prefix: No
 Proxy Redundancy Method: Normal

Proxy and Registration			
Proxy:	10.10.10.50		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	yes
Register:	no	Make Call Without Reg:	yes
Register Expires:	3600	Ans Call Without Reg:	yes
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal

Dial Plans

What we are doing in this section is telling the SPA-3102 how to route a call when one comes in. Basically the dialplan we use is going to say, "go to the IP address I specify and tell the system the phone number I am giving you", this will also allow you to use the phone number you specify as a DID for inbound route control.

Dial Plan 1: (S0<: 7145551212@10.10.10.50 :5060>)

The 7145551212 is YOUR phone number, change this to be the phone number people dial to reach you.
 The 10.10.10.50 is the IP address of your PBX, change this to match your system configuration.

Dial Plans	
Dial Plan 1:	(S0<:7145551212@10.10.10.50:5060>)
Dial Plan 2:	(soc.)
Dial Plan 3:	(soc.)
Dial Plan 4:	(soc.)
Dial Plan 5:	(soc.)
Dial Plan 6:	(soc.)
Dial Plan 7:	(soc.)
Dial Plan 8:	(soc.)

VOIP-To-PSTN Gateway

This section of the configuration tells the SPA-3102 how to handle calls that come from your PBX and go out to the phone line. The factory default settings should work properly but again, I will give you the working settings just in case something isn't set right:

VoIP-To-PSTN Gateway Enable: Yes
 VoIP PIN Max Retry: 3
 Line 1 VoIP Caller DP: 3
 Line 1 Fallback DP: 1
 VoIP Caller Auth Method: None
 One Stage Dialing: Yes
 VoIP Caller Default DP: 1

VoIP-To-PSTN Gateway Setup			
VoIP-To-PSTN Gateway Enable:	yes	VoIP Caller Auth Method:	none
VoIP PIN Max Retry:	3	One Stage Dialing:	yes
Line 1 VoIP Caller DP:	1	VoIP Caller Default DP:	1
Line 1 Fallback DP:	none		

PSTN-To-VOIP Gateway

This section tells the SPA-3102 how to handle calls from the phone line to the PBX. While the default settings will mostly work,

there are two changes you will most likely want to make, again, here is the complete list:

PSTN-To-VoIP Gateway Enable: Yes

PSTN Ring Thru Line 1: No

PSTN CID For VoIP CID: Yes

PSTN Caller Default DP: 1

Line 1 Signal Hook Flash To PSTN: Disabled

PSTN Caller Auth Method: none

PSTN PIN Max Retry: 3

PSTN CID Number Prefix: <leave this blank>

Off Hook While Calling VoIP: No

PSTN CID Name Prefix: <leave this blank>

PSTN-To-VoIP Gateway Setup			
PSTN-To-VoIP Gateway Enable:	yes	PSTN Caller Auth Method:	none
PSTN Ring Thru Line 1:	no	PSTN PIN Max Retry:	3
PSTN CID For VoIP CID:	yes	PSTN CID Number Prefix:	
PSTN Caller Default DP:	1	Off Hook While Calling VoIP:	no
Line 1 Signal Hook Flash To PSTN:	Disabled	PSTN CID Name Prefix:	

Save your settings and your SPA-3102 should be ready to go!

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