Cisco SPA112 and SPA 122 Basic Configuration Guide

by Wahroonga Farm

The Cisco SPA112 and SPA122

Cisco's SPA1XX series are the next generation successors to the popular but now discontinued, Linksys (Cisco) PAP2 and PAP2T VoIP ATAs.

From a configuration point of view, very little has changed. If you can configure a PAP2(T) you can configure a Cisco SPA112 or SPA122 and vice versa.

The Cisco SPA112 or SPA122 are two port ATAs (Analogue Telephone Adapters) with interfaces for two analogue phones. Each port may be configured with the same or different SIP VoIP accounts, or services. For example one VoIP service provider may give great National call rates, whilst another great International rates. You can connect a telephone for each enabled line, or you could configure one as a voice, and the other as a FAX line.

Both ATA's use the same firmware versions e.g Version 1.3.3(015) for SPA112 and SPA122 so VoIP features and functionality are identical. The difference between the two is the hardware NAT/DHCP router in the Cisco SPA122. You may disable the router via software configuration (see below ¹) and the devices become identical. Read more here.

Connecting the ATA

Follow the Cisco handbook directions for connection details.
Determine the ATA WAN port IP address

Use your routers DHCP client list to determine the ATA’s IP address (how to here) or:

- Connect an analogue phone to phone 1 of the ATA.
- On the phone dial * * * *
- After the greeting plays; dial 1 1 0 #
- Write down the address.
- Hang up the telephone.

Note: The SPA122 may also be accessed via the Ethernet port using 192.168.15.1. The SPA112 must be accessed via the IP address issued by your router.

Configuring the SPA100 series

Before you start, download the latest firmware from the Cisco website.

Open a browser session and type in the IP address obtained via the preceding step e.g. 192.168.1.130.

Use admin/ admin to log in. To modify the password, see below.
Move to the administration screen, check your firmware version and upgrade as necessary.

Set Time Zone

Should you wish to disable the SPA122 router functionality, move to Network Setup and set Networking Service to Bridge, followed by submit.
Quick Setup

Select the Quick Setup screen and fill in the account details as shown below.

Dial Plans are VSP specific.

Telecube recommends: (*xx|0[0-9]xxxxS0|0[234]78|xxxxxxxxS0|[3-9]xxxxxxxxS0|1831[23478]xxxxxxxS0|1831[3-9]xxxxxxS0|1800xxxxxxS0|1300xxxxxxS0|13[1-9]xxxxS0|0198xxxxxxS0|0011xx.|12x|19xx|x|x.) *Note: Telecube currently does not support 000.*

MyNetFone recommends: (*xx|000S0|121S0|151S0|181S0|[2-9]xxxxxxS0|0[23478]xxxxxxxS0|0011xxx.|1800xxxxxxS0|1300xxxxxxS0|13[1-9]xxxxS0|xxx.)

The dial plan improves post dial delay and screens for disallowed number sequences. You may experiment with different dial plans. The SPA112 default setting is: (*xx|[3469]11|0|00|[2-9]xxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.). Some useful reading on dial plans [here](#).

Select Submit to save the changes
At this stage, your ATA should successfully register and connect line 1.

Regional Settings

Go to Voice Regional settings and configure the following settings. Whilst not essential for operation, these changes will make your handset ‘sound’ more Australian. Thanks to Jason for these settings.

Under Call Progress Tones change the following:

- **Dial tone:** 400@-19,425@-19,450@-19;10(*/0/1+2+3)
- **Busy Tone:** 425@-19;10(.375/.375/1)
- **Reorder Tone:** 425@-19, 425@-29;60(.375/.375/1,.375/.375/2)
- **Ring Back Tone:** 400@-19,425@-19,450@-19;*(.4/.2/1+2+3,.4/2/1+2+3)
- **MWI Dial Tone:** 400@-19,425@-19,450@-19;2(.1/.1/1+2);10(*/0/1+2)
Under the *Distinctive Ring Patterns*, set *Ring1 Cadence*:

60(.4/2,.4/2)
Nothing changes here as we scroll down

Under **Miscellaneous** change **FXS Port Impedance** by selecting the appropriate value:

**FXS Port Impedance:** $220 + 820 || 115\text{nF}$

Select **Submit** to save the changes.
Line 1 Configuration

Go to Voice then Line 1.

Set NAT Mapping Enable and NAT Keep Alive Enable to Yes

If you are using an nbn Satellite, Wireless or Fibre service and TC-1 is enabled by your provider (more about this below), you must change the SIP ToS/DiffServ Value from 0x68 to 0xb8, to ensure voice continuity.

Note: DSCP is normally defined as decimal, with the 2 least significant bits of the TOS byte discarded. 0xb8 (decimal 184) becomes decimal 46. The NBN Fibre technical specifications list TC-1 as requiring DHCP (decimal) 40-47. It appears to be the same as satellite.

If TC-1 is not enabled, leave as the default configuration.

Note: The nbn TC-1 reserves 0.15 Mbps (150KB) for VoIP traffic and the VoIP call will be protected from congestion within the nbn network and hopefully beyond.

![Phone Adapter Configuration Utility](image-url)
Scroll down. Nothing to change here.

We have already configured most of this VoIP provider specific information at the Quick Setup screen.

At Register Expires set 120 (may reduce call drop-outs). 180 is the generally recommended setting.
Nothing to change here.

Audio configuration and Dial Plan

If you are using a sound ADSL or mobile service or NBN fixed line, wireless or Sky Muster connection the default Preferred Codec ie G711u is best. If you are using expensive data or a poor data connection you might try G729a. This codec may provide a more reliable service. It’s a ‘case of suck it and see’. This setting should match your VSP codec configuration/priority.
Select Submit to save the changes. Your service should now be fully functional.

Comparing G711A to G729A

Voice quality is much improved with G711a (much greater dynamic range), but clarity is still fine with G729a.

The charts below display average bandwidth utilisations as recorded by an OpenWRT USB router with a Sierra 320U modem providing the 3G service:

G729a – 23.11 kbits/s inbound, 30.66 kbits/s outbound  
G711a – 80.61 kbits/s inbound, 87.40 kbits/s outbound

The G711a codec uses a bit over three times the bandwidth utilisation of the G729A codec.
**Additional NBNCo satellite service tips**¹

Depending on your NBN service configuration, you may require a router that supports SIP-ALG.

Note: This tip is likely no longer required. It was necessary for the *nbn* Interim Satellite Service (ISS).

If you request (for best satellite VoIP performance) an NBN Interim Satellite Service Traffic Class 1 (TC-1) service (60/60K reserved) channel; you must:

1. Have a router that supports SIP-ALG,
   or
2. configure a STUN server in your ATA

Without SIP-ALG enabled the speech path will be broken. Few consumer wireless routers offer this feature although many ADSL routers do. Wireless routers that support SIP-ALG (my list):

- Current [Linksys](https://www.linksys.com) wireless routers.
- [Netcomm NB16WV](https://www.netcomm.com.au/products/internet-access-modem-router/nb16wv) – an old 10/100 ADSL modem/router/VoIP with a WAN port. Not a recommended device; but you may have one lying around and it’s worth a try.
- [TP-Link C2](https://www.tp-link.com) - yet to be released (my choice).
- [TP-Link TD-VG3631](https://www.tp-link.com) - 10/100 ADSL router with a WAN port.
Configuring a TC-1 NBN Co ISS service ATA with a STUN server

If your router does not support SIP-ALG then there is a work around.

Move to the Voice then SIP screen.

   Set STUN Enable to Yes

   Set STUN Server to stun2.faktortel.com.au

Submit

Your ATA will access a STUN server at VoIP provider FactorTel, overcoming the router SIP ALG limitations.
And a very warm thank you to Whirlpoolers

Paul Rees (SkyMesh), Robnll, Finite State Machine and Monkeymind,

for their able and willing assistance
**Addendum**

**Password Configuration**³

Move to Administration >Management >User List page to manage the two user accounts for the password configuration utility.

The administrator-level account has the default username admin and password admin. The user-level account has access to modify a limited set of features. This account has the default username cisco and password cisco. To update a password:

**STEP 1** In the User List table, click the pencil icon for the account that you want to update.

**STEP 2** On the User Account page, enter the username and password, as described below.

- **Username:** Enter a username.
- **Old Password (administrator account only):** Enter the existing password. The default administrator password is admin. The default guest password is cisco.
- **New Password:** Enter up to 32 characters for your new password.
- **Confirm New Password:** Enter the new password again, to confirm.
STEP 3  After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Router DHCP client list

Here is a screen snap of a TP-Link WDR3600’s DHCP client list showing the IP address of the ATA as 192.168.0.101.

![DHCP Clients List](image)

<table>
<thead>
<tr>
<th>ID</th>
<th>Client Name</th>
<th>MAC Address</th>
<th>Assigned IP</th>
<th>Lease Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>ancast-</td>
<td>01-01-01-01</td>
<td>192.168.0.100</td>
<td>01:25:47</td>
</tr>
<tr>
<td>2</td>
<td>SPA122</td>
<td>01:02:03:04</td>
<td>192.168.0.101</td>
<td>01:35:25</td>
</tr>
<tr>
<td>3</td>
<td>android-</td>
<td>05-06-07-08</td>
<td>192.168.0.102</td>
<td>01:25:51</td>
</tr>
<tr>
<td>4</td>
<td>ancast-</td>
<td>09-10-11-12</td>
<td>192.168.0.103</td>
<td>01:20:10</td>
</tr>
<tr>
<td>5</td>
<td>John17</td>
<td>13-14-15-16</td>
<td>192.168.0.104</td>
<td>01:19:57</td>
</tr>
<tr>
<td>6</td>
<td>SuePC</td>
<td>17-18-19-20</td>
<td>192.168.0.105</td>
<td>01:45:19</td>
</tr>
</tbody>
</table>

Most routers will have a similar feature.

A general Voip Codec description (not Cisco ATA specific)

**G.722**

Excellent voice quality. The **broadband** speech codec **G.722** works at the same bit rate as G.711 (64 kbit/s per speech connection) but with a higher sampling rate. This allows higher frequencies to be played back. The speech tone is therefore clearer and better than for the other codecs (High Definition Sound Performance).

**G.711 a law / G.711 µ law**

Excellent voice quality (comparable with ISDN). The necessary bandwidth is 64 kbit/s per voice connection.

**G.726**

Good voice quality (inferior to that with G.711 but better than with G.729). Your phone supports G.726 with a transmission rate of 32 kbit/s per voice connection.

**G.729**

Average voice quality. The necessary bandwidth is less than or equal to 8 kbit/s per voice connection.

To save additional bandwidth and transmission capacity, on VoIP connections that use the G.729 codec you can suppress the transmission of voice packets in pauses ("silence suppression"). Instead of the background noises in your environment, your caller then hears a synthetic noise generated in the receiver (option: **Enable Annex B for codec G.729**).