

# Linksys SPA-922 Configuration Guide



The following instructions are for the Linksys SPA922 phone but can be used a basis for configuring other Linksys VoIP phone such as the SPA942 etc.

First we always recommend loading the latest firmware onto the phone if necessary:

- Visit <http://www.linksys.co.nz>
- Click on Downloads
- Select VoIP Phones from drop down menu under the VoIP category
- Select SPA922 from the drop down menu under the picture that looks like your phone
- Select Version 1.0
- Click on firmware to download the Firmware ZIP file
- Open the ZIP file in a ZIP utility such as WinZip
- Click Extract and extract your files to a folder you choose (e.g. C:\SPA922)
- Use Windows Explorer to open a window on your selected folder
- Click on the .exe Application
- Accept the warning and Click Continue
- Enter the IP Address of the SPA922 above into the Window and click OK
- Click Upgrade to update your SPA922 firmware to the new version you have downloaded
- Wait until the progress indicator has finished doing the upgrade. Do NOT unplug your SPA922 during the upgrade.
- When successfully upgraded you should see a message like 'Your SPA has been successfully upgraded to the version shown below' (Latest version at time of writing was 5.2.8)
- Now in your web browser put in the SPA922 IP Address again and on the Info page you should see the new firmware version displayed beside 'Software Version'

## System Tab

These settings are up to you and your network but I recommend you password protect the web interface to stop users messing with settings:

- Enable web server = Yes
- Enable web Admin access = Yes
- Admin passwd: <your admin password here>
- User passwd: <your admin password here>
- Connection Type: DHCP
- Primary DNS: 202.180.76.161
- Secondary DNS: 202.180.76.162
- Primary NTP server: ntp.orca.com.co.nz

The rest should be okay as the defaults

## SIP Tab

Change the following SIP Timer Values:

- Reg Max Expires = 600
- Reg Retry Intvl = 10
- Reg Retry Long Intvl = 20

This should allow the phone to recover from Internet issues more quickly and give MOS info in BYE messages (handy for troubleshooting)

RTP Parameters

- Stats In BYE: yes

Change the following NAT Support Parameters (optional):

- STUN Enable: yes
- STUN Server: stun.OrcaCom.co.nz



STUN is not strictly needed but there should be no harm in enabling it.

### Provisioning Tab

You can leave auto-provisioning enabled if you like. No harm in leaving this turned on for now even if you are going to do manual provisioning.

### Regional Tab

If you are really keen then you can set the tones to standard NZ tones instead of the defaults. I don't usually bother and stick with the default tones, but some people prefer to replicate Telecom NZ completely right down to the Dial/Busy Tones etc. It's up to you. Here are the NZ tones anyway:

- Call Progress Tones
  - Dial Tone: 400@-9;30(\*0/1)
  - Outside Dial Tone: 420@-16;10(\*0/1)
  - Prompt Tone: 520@-19,620@-19;10(\*0/1+2)
  - Busy Tone: 400@-9;\*(.5/.5/1)
  - Reorder Tone: 400@-9;15(.25/.25/1+2)
  - Off Hook Warning Tone: 400@-10,680@0;\*(.125/.125/1+2)
  - Ring Back Tone: 400@-19,450@-19;\*(.4/.2/1+2,.4/.2/1+2,2/0/0)
  - Confirm Tone: 600@-16;1(.25/.25/1)
  - SIT1 Tone: 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
  - SIT2 Tone: 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
  - SIT3 Tone: 914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
  - SIT4 Tone: 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)
  - MWI Dial Tone: 400@-19;2(.1/.1/1);28(\*0/1)
  - Cfdw Dial Tone: 350@-19,440@-19;2(.2/.2/1+2);10(\*0/1+2)
  - Holding Tone: 600@-19;\*(.1/.1/1,.1/.1/1,.1/9.5/1)
  - Conference Tone: 350@-19;20(.1/.1/1,.1/9.7/1)
  - Secure Call Indication Tone: 397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2)
- Distinctive Ring Patterns
  - Cadence 1: 60(.4/.2,.4/2)
  - Cadence 2: 60(.3/.2,1/.2,.3/4)
  - Cadence 3: 60(.8/.4,.8/4)
  - Cadence 4: 60(.4/.2,.3/.2,.8/4)
  - Cadence 5: 60(.2/.2,.2/.2,.2/.2,1/4)
  - Cadence 6: 60(.2/.4,.2/.4,.2/4)
  - Cadence 7: 60(.4/.2,.4/.2,.4/4)
  - Cadence 8: 60(0.25/9.75)

### Miscellaneous

- Set Local Time (HH/mm): (Set current hour and minute)
- Time Zone: GMT+12
- Daylight Saving Time Rule: start=4/1/7/3:0:0;end=9/-1/7/2:0:0;save=-1
- DTMF Playback Level: 0
- DTMF Playback Length: .1

### Phone Tab

#### General

- Station Name: <phone number or whatever you want>
- Voice Mail Number: \*55

#### Supplementary Services

If you wish to use OrcaCom's versions of these features rather than those built into the phone (recommended) then you should turn off the following features

- DND Serv: no

- Block CID Serv: no
- Secure Call Serv: no
- C fwd all Serv: no
- C fwd busy Serv: no
- C fwd No Ans Serv: no



## Ext1 Tab

### General

- Line Enable: yes

### Network Settings

- SIP TOS/Diffserv Value: 0x1A (up to your network, this is what OrcaCom use)
- SIP Cos Value: 3 (up to your network, this is what OrcaCom use)
- RTP TOS/DiffServ Value: 0x2E (up to your network, this is what OrcaCom use)
- RTP CoS Value: 6 (up to your network, this is what OrcaCom use)
- Network Jitter Level: very high (this depends on the latency/jitter of your network through to OrcaCom. Very high should be fine, but can be increased/decreased for network conditions)
- Jitter Buffer Adjustment: up and down

### SIP Settings

- SIP Port: 5060 (*\*see troubleshooting below if you are having problems*)

### Call Feature Settings

- Blind Attn-Xfer Enable: yes
- Message Waiting: yes
- Voice Mail Server: <phone number>

### Proxy and Registration

- Proxy: sip.orcacom.co.nz (*\*see troubleshooting below if you are having problems*)
- Use Outbound Proxy: yes
- Outbound Proxy: sip.orcacom.co.nz (*\*see troubleshooting below if you are having problems*)
- Use OB Proxy In Dialog: yes
- Register: yes
- Make Call Without Reg: yes
- Register Expires: 300
- Ans Call Without Reg: yes
- Use DNS SRV: yes

### Subscriber Information

- Display Name: <Your Name> (e.g. Fred Bloggs)
- User ID: <Your Phone Number> (e.g. 099749000)
- Password: <Your password>
- Use Auth ID: yes
- Auth ID: <Your Phone Number> (e.g. 099749000)

### Audio Configuration

- Preferred Codec: G711a (unless you don't have much bandwidth - in which case select G729)
- Use Pref Codec only: no
- G729a Enable: yes
- DTMF Process AVT: yes
- Silence Supp Enable: no
- DTMF Tx Method: AVT



## Dial Plan

- Dial Plan: (0[2-9][2-9]xxxxxx[2-9]xxxxxx90[1-8])021x.|0508x.|0800x.|025x.|027x.|028x.|029x.|00x.|1xx|\*x.)

## User Tab

### Supplementary Services

- CW Setting: yes
- Block CID Setting: no

Most other settings are up to you and your preferences. We have only highlighted the ones that are important.

## Troubleshooting

If you are having problems connecting to OrcaCom or have problems with calls connecting/dropping or one-way audio then you may have a router/firewall device which is 'interfering' with the 'SIP' traffic causing problems for your phone. You can try and use an alternate port number on OrcaCom to get around this by changing the following settings in your configuration:

### SIP Settings

- SIP Port: 50600

### Proxy and Registration

- Proxy: sip.orcacom.co.nz:50600
- Outbound Proxy: orcacom.co.nz:50600

This will cause the port number used for SIP traffic on both the phone and out to OrcaCom to be '50600' instead of '5060' which is often enough to resolve the problems with interfering routers.