Sipura SPA-3000
Simplified Users Guide
Version 1.1b

A Step by Step Introduction

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Section 1 – Getting Started

When you receive your SPA-3000 it should contains the following items:

1. SPA-3000
2. Power supply
3. Phone cable
4. Network cable
5. Quickstart guide

Setting up the hardware:

Step 1: Connect the power cable to your SPA-3000
Step 2: Connect the network cable from your SPA-3000 to your DHCP enabled modem/router
Step 3: Connect an analogue phone to the "Phone" port in the SPA-3000.
Step 4: Turn the power on
Step 5: Pick up the phone and dial **** (You should hear a voice saying Sipura Configuration Menu)
Step 6: Dial 110# (Write down the IP address that is returned)
Step 7: Connect your PSTN line to the "Line" port of the SPA-3000.

At this stage, you can choose whether to upgrade the firmware. The latest version of the SPA-3000 firmware at the time of writing this document is 3.1.7g.

To upgrade the firmware see **Appendix D**.

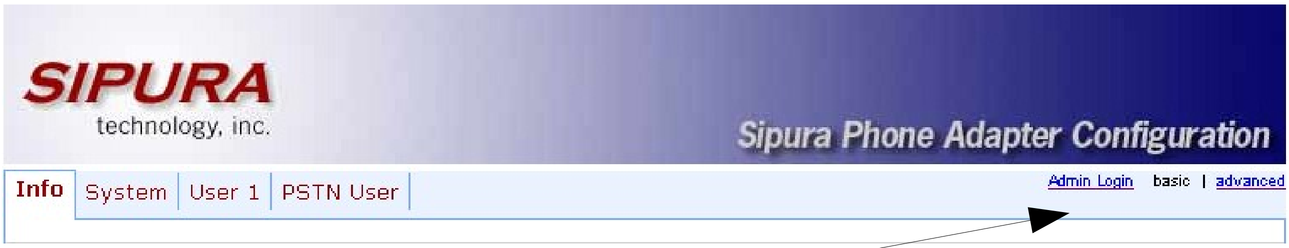
Section 2 – Configuring the SPA-3000 via the Web Interface

Once you have the IP address of your SPA-3000 you can access the web interface by opening a web browser and typing the following in the Address :

http://<*IP Address of SPA-3000*>

For example: http://192.168.1.10

You should now see the Sipura SPA-3000 Phone Adapter Configuration screen. Now you will need to log into the unit as Administrator and change the view to advanced. See images below.



Click on "Admin Login"

Click on "advanced"

Note: You can also go directly to the admin/advanced section by typing

http://192.168.1.10/admin/advanced

Section 3 – Regional Settings

Click on the "Regional" tab.

The following changes are optional, however they will make your SPA-3000 sound more Australian. Under the Call Progress Tones and Distinctive Ring Patterns headings change:

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)
Ring1 Cadence:	60(.4/.2,.4/2)

Call Progress Tones	
Dial Tone:	400@-19,425@-19,450@-19;10(*/0/1+2+3)
Second Dial Tone:	420@-19,520@-19;10(*/0/1+2)
Outside Dial Tone:	420@-16;10(*/0/1)
Prompt Tone:	520@-19,620@-19;10(*/0/1+2)
Busy Tone:	425@-19;10(.375/.375/1)
Reorder Tone:	425@-19, 425@-29;60(.375/.375/1,.375/.375/2)
Off Hook Warning Tone:	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone:	400@-19,425@-19,450@-19;*(.4/.2/1+ 2+3,.4/2/1+2+3)
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@-19,425@-19,450@-19;2(.1/.1/1+2); 10(*/0/1+2)
Cfwd 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)
VoIP PIN Tone:	600@-10;*(0/1/1,.1/.1/1,.1/.1/1,.1/.5/1)
PSTN PIN Tone:	600@-10;*(0/.7/1,.2/.1/1,.2/.1/1,.2/.5/1)
PSTN Warning Tone:	600@-10;5(0/.5/1,.05/.05/1,.05/.7/1)
Feature Invocation Tone:	350@-16;*(.1/.1/1)

Distinctive Ring Patterns			
Ring1 Cadence:	60(.4/.2,.4/2)	Ring2 Cadence:	60(.3/.2,1/.2,.3/4)
Ring3 Cadence:	60(.8/.4,.8/4)	Ring4 Cadence:	60(.4/.2,.3/.2,.8/4)
Ring5 Cadence:	60(.2/.2,.2/.2,.2/.2,1/4	Ring6 Cadence:	60(.2/.4,.2/.4,.2/4)
Ring7 Cadence:	60(.4/.2,.4/.2,.4/4)	Ring8 Cadence:	60(0.25/9.75)

The following changes are required under the Miscellaneous heading :

FXS Port Impedance: 220+820||115nF

Miscellaneous			
Set Local Date (mm/dd):		Set Local Time (HH/mm):	
Time Zone:	GMT+11:00	FXS Port Impedance:	220+820||115nF
Daylight Saving Time Rule:	start=3/-1/7/3;end=10/-1/7/2;save=-1		
FXS Port Input Gain:	-3	FXS Port Output Gain:	-3
DTMF Playback Level:	-16	DTMF Playback Length:	.1
Detect ABCD:	yes	Playback ABCD:	yes
Caller ID Method:	Bellcore(N.Amer,China)	FXS Port Power Limit:	3
Caller ID FSK Standard:	bell 202	Feature Invocation Method:	Default

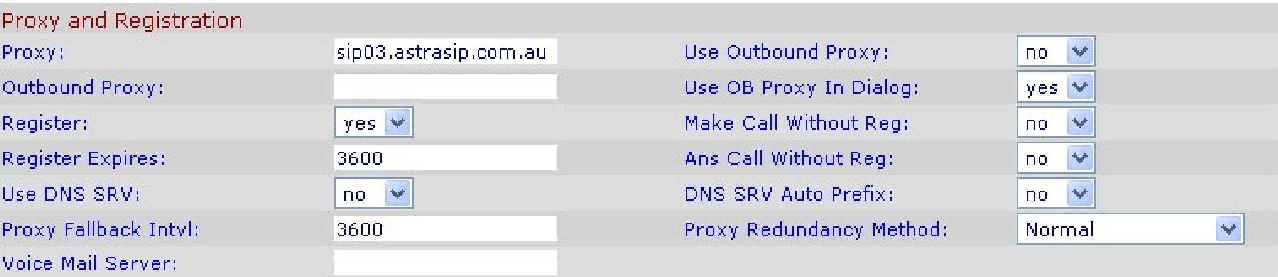
If you would like to configure the Time and Daylight savings, see **Appendix F**. This is completely optional.

Section 4 – Line 1 Settings

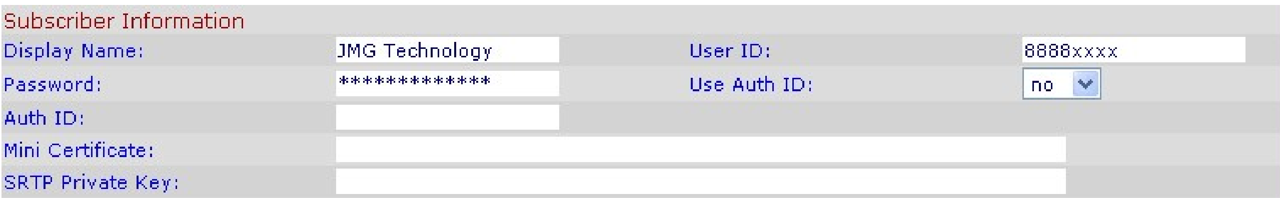
Click on the "Line 1" tab.

The first thing that needs to be changed is under the Proxy and Registration heading. Here you will need to enter the proxy and registration information given to you by your VoIP provider.

The following image shows the setup for Astratel.



The second thing that needs to be configured is under the Subscriber Information heading. Here you will need to enter your user id and password for given to you from your VoIP provider.



For **some** VoIP providers you will also have to enter the Auth ID and set Use Auth ID to yes.

The next thing that needs to be altered is under the Audio Configuration heading. The following change is required.

Preferred Codec: G729a

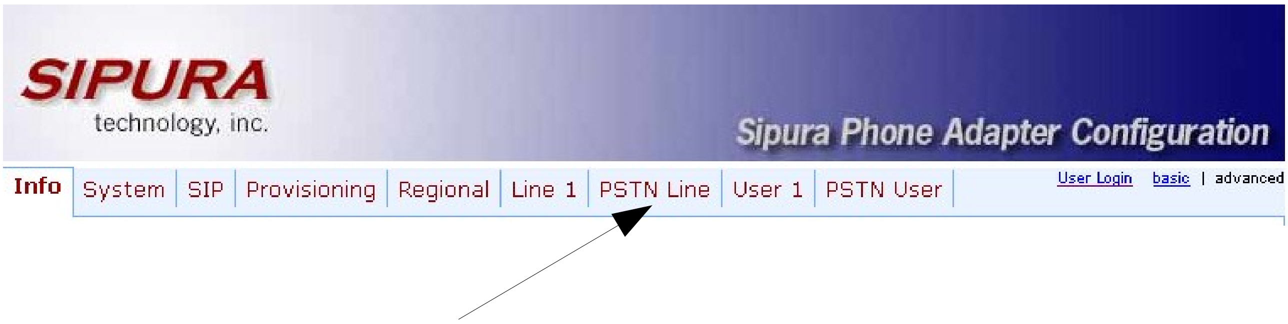
Audio Configuration			
Preferred Codec:	G729a	Silence Supp Enable:	no
Use Pref Codec Only:	no	Silence Threshold:	medium
G729a Enable:	yes	Echo Canc Enable:	yes
G723 Enable:	yes	Echo Canc Adapt Enable:	yes
G726-16 Enable:	yes	Echo Supp Enable:	yes
G726-24 Enable:	yes	FAX CED Detect Enable:	yes
G726-32 Enable:	yes	FAX CNG Detect Enable:	yes
G726-40 Enable:	yes	FAX Passthru Codec:	G711u
DTMF Process INFO:	yes	FAX Codec Symmetric:	yes
DTMF Process AVT:	yes	FAX Passthru Method:	NSE
DTMF Tx Method:	Auto	FAX Process NSE:	yes
Hook Flash Tx Method:	None	FAX Disable ECAN:	no
Release Unused Codec:	yes	Symmetric RTP:	yes

The final item that requires changing is under the Dial Plan heading. The dial plan will vary from provider to provider. The dial plan show below is an example only, and may require alteration depending on which VoIP provider you subscribe to. See **Appendix A** for a detailed description of how dial plans work.

Dial Plan: (000S0<:@gw0>|<#0,:>xx.<:@gw0>|xx.)

Dial Plan	
Dial Plan:	(000S0<:@gw0>|<#0,:>xx.<:@gw0>|xx.)
Enable IP Dialing:	no

Section 5 – PSTN Line Settings



Click on the "PSTN Line" tab.

The first thing that needs to be changed is under the PSTN Disconnect Detection heading. The following change is required:

Disconnect Tone: 425@-30,425@-30;10(.375/.375/1+2)

PSTN Disconnect Detection			
Detect CPC:	yes	Detect Polarity Reversal:	yes
Detect PSTN Long Silence:	no	Detect VoIP Long Silence:	no
PSTN Long Silence Duration:	30	VoIP Long Silence Duration:	30
PSTN Silence Threshold:	medium	Min CPC Duration:	0.2
Detect Disconnect Tone:	yes		
Disconnect Tone:	425@-30,425@-30;10(.375/.375/1+2)		

The next thing that needs to be changed is under the International Control heading. The following changes are required:

FXO Port Impedance: 220+820||120nF
PSTN To SPA Gain: 3
On-Hook Speed: 26ms (Australia)

International Control			
FXO Port Impedance:	220+820||120nF	Ring Frequency Min:	10
SPA To PSTN Gain:	0	Ring Frequency Max:	100
PSTN To SPA Gain:	3	Ring Validation Time:	256 ms
Tip/Ring Voltage Adjust:	3.5 V	Ring Indication Delay:	512 ms
Operational Loop Current Min:	10 mA	Ring Timeout:	640 ms
On-Hook Speed:	26 ms (Australia)	Ring Threshold:	13.5-16.5 Vrms
Current Limiting Enable:	no	Ringer Impedance:	High (Normal)
Line-In-Use Voltage:	30		

Appendix A – Dial Plans

Dial plans can be very confusing at first glance. However they are invaluable feature of the SPA-3000 so you should at least learn the basics of how they work.

Dial Plan Syntax

( )	- The entire dial plan must be surrounded by an open and close bracket.
|	- Each individual dial plan must be separated by a pipe | character.
0-9	- Treated as normal digits
x	- Treated as any normal digit 0-9 on phone
*****	- Treated as normal * character on phone
#	- Treated as normal # character on phone
.	- Repetition
< : >	- Replacement, eg <02:612> means replace 02 with 612
<:@gw0>	- Gateway 0 is the PSTN line
<:@gw1>	- Gateway 1 (Advanced Feature)
<:@gw2>	- Gateway 2 (Advanced Feature)
<:@gw3>	- Gateway 3 (Advanced Feature)
<:@gw4>	- Gateway 4 (Advanced Feature)
S0	- Dial Immediately
!	- Barring a number, place this at the end of the number to bar it
,	- Provides a dial tone
[ ]	- Limiting choices, eg [24] means either 2 or 4, [2-5] means 2,3,4 and 5, [24-68] means 2,4,5,6,8

Example Dial Plans

Dial Plan 1: (000S0<:@gw0>)

Description: The above dial plan is extremely simple, yet extremely important. When you dial 000 (Emergency number) your call will go out through Gateway 0 (<:@gw0) which is your normal PSTN line, immediately (S0) after you have dialed the 3^{rd} 0.

Dial Plan 2: (000S0<:@gw0>|1800xxxxxxS0<:@gw0>)

Description: The above dial plan contains two individual plans, building on from Dial Plan 1. You will notice that a | separates the 1^{st} dial plan from the 2^{nd} .The 2^{nd} dial plan is used to route 1800 numbers through the your PSTN line. It works the same way as the 1^{st} dial plan, in that when you dial a 1800 number followed by 6 other digits (0-9) it will be directed through your PSTN line.

Dial Plan 3: (<*1:0123456789>)

Description: This plan demonstrates replacement. If you dial a * followed by a 1 then the number 0123456789 would be dialed.

Dial Plan 4: (<0:61>[2-9]xxxxxxxxS0)

Description: This plan demonstrates replacement and limiting choices. When you dial an 0 followed by a 2,3,4,5,6,7,8 or 9 and then nine of any other digit (0-9) it will prepend 61 and remove the 0. So if you rang 02 123456789 the actual number that would be called would be 61 2 123456789.

Dial Plan 5: (1900xxxxxx!)

Description: This plan demonstrates number barring. If you enter a 1900 followed by 6 more digits (0-9) you call will not be placed.

Dial Plan 6: (<#9:>xx.<:@gw0>)

Description: This plan demonstrates replacement and repetition. When you enter a #9 followed by any number of digits(a timeout is used to determine the end) it will go out through the PSTN line (Gateway 0).

Putting it all together

Dial Plan 7: (000S0<:@gw0>|1800xxxxxxS0<:@gw0>|1300xxxxS0<:@gw0>|
1900xxxxxx!| 0[2-9]xxxxxxxxS0|<#9:>xx.<:@gw0>)

Description: This plan combines elements from all the above dial plans. It routes all 000, 1800, 1300 calls out via the PSTN line. Its bars 1900 numbers. It allows an Australian land line to be called and it also allows you to select the PSTN line by dialing a #9.

Appendix B - Factory Reset

To perform a factory reset on your SPA-3000 remove the Ethernet cable and the PSTN line cable, leaving just the power and the phone connected. Dial **** on the phone. You should hear a Sipura message asking you to enter your selection. . Then dial 73738#.

WARNING: This will restore your unit back to factory defaults, all your information will be lost.

Appendix C - Saving your Configuration

Method 1 – ProgramUtility

There is a utility that has been written to save/restore SPA-3000 configurations. The file is called NewSipuraUtil and can be downloaded from the following site:

http://www.dualarrow.com

Method 2 - Manually

To save your configuration, log into your SPA-3000 web interface as admin. Change the view to advanced. Select the ***File – Save As*** option from your web brower and save the configuration page to your PC's hard drive. Thats it!

Now to restore settings that you have previously saved, you need to edit the configuration page that you saved to your PC's hard drive. To do this, find the page on your hard drive, right click on it and ***Open With – Notepad***. Now, do a search for the following line of code.

<FORM action=”asipua.spa” ....

You need to change this line to read:

<FORM action=”http://*IP Address of Sipura*/admin/asipura.spa” ....

Where *IP Address of Sipura* is the IP address of your Sipura SPA-3000.

Now save the page, then load it up in your web browser, when you hit Submit Changes, your saved configuration will be loaded back into your SPA-3000.

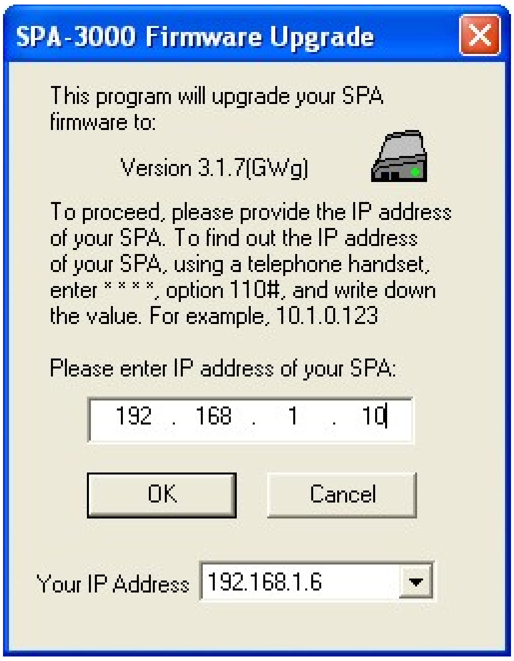
Appendix D – Upgrading Firmware

The latest firmware for the Sipura SPA-3000 can be located on the Sipura support site.

http://www.sipura.com/support/index.htm

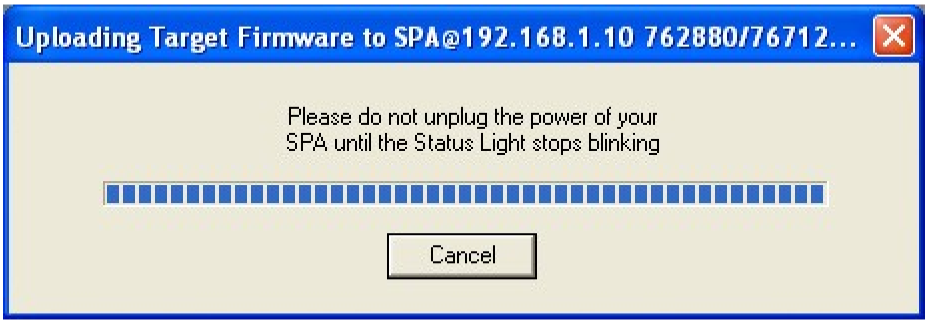
To upgrade the firmware in your SPA-3000 download the latest firmware, unzip it and run the exe file provided. At the time of writing this document v3.1.7Gwg is the latest firmware,

The first thing that you need to do is enter the IP address of your SPA into the spaces provided. In the example below the SPA-3000 is located at 192.168.1.10.



Click on the OK button to begin so the program can find your SPA-3000. When the program locates the SPA, it will interrogate it and you will be shown a confirmation screen.

Click on the Upgrade button to begin the upgrade procedure. The upgrade itself should only take a minute or so. Whatever you do don't turn the power of during this operation.



Once the procedure has finished, you should be presented with the following screen and your upgrade has been completed.

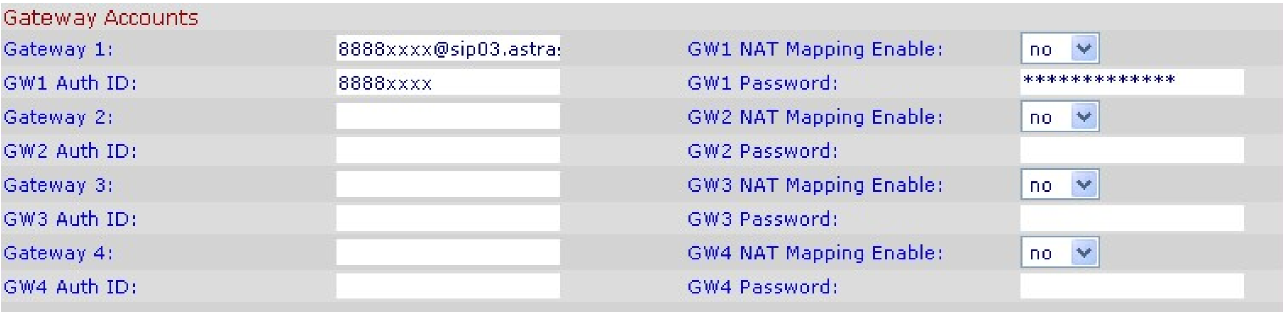
Note: The upgrade procedure does not effect your current settings.

Appendix E – Setting up Gateways

The SPA-3000 allows you to configure up to 4 extra VoIP providers, through the use of gateways. Please note that not all providers can be used in the extra gateways because they require registering.

To add a provider into one of the 4 gateways you will need to know user userid, password and the proxy address of the VoIP provider.

The following image shows how to set up Astratel as gateway 1.



Setup Gateway 1.

Gateway 1:	*userid@proxy*
GW1 Auth ID:	*userid*
GW1 Password:	*password*

where *userid* is your VoIP number given to you by your provider, *proxy* is the sip proxy address and *password* is the password given to you by your provider.

The next thing that is required is to add an entry into the dial plan to allow calls to be made using the gateway you have just setup.

For simplicity sake, I have added the following entry to the dial plan.

<#1,:>xx.<:@gw1>

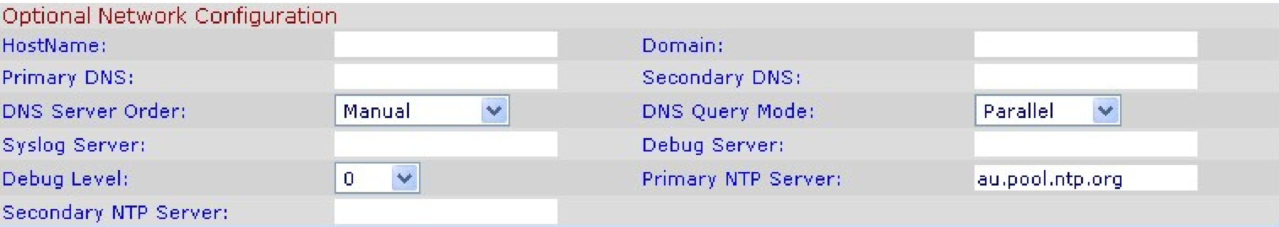
What this means is any number you dial after typing #1 will go out through gateway 1.

You can setup the other 3 gateways in exactly the same way. Gateway 2 uses <:@gw2>, Gateway 3 uses <:@gw3> and Gateway 4 uses <:@gw4>

Appendix F – Setting up a Time Server and Daylight Saving Rules

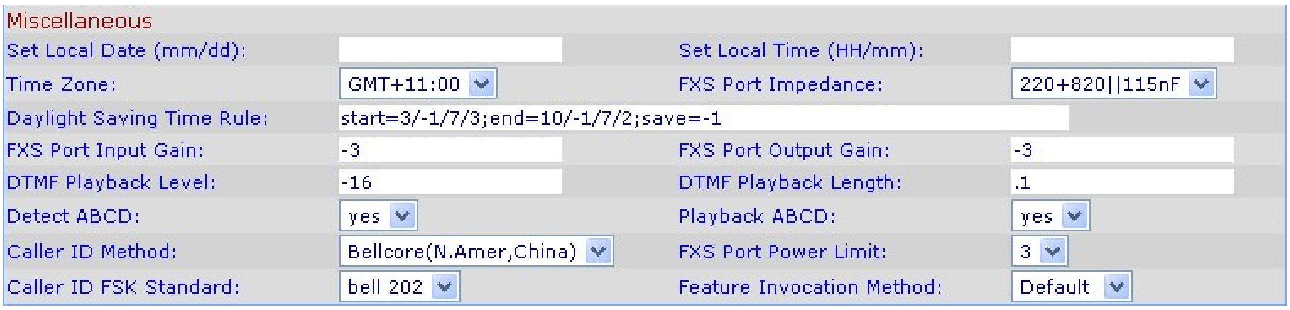
You can configure the SPA-3000 to automatically retrieve the current date and time. (Including daylight savings). To do this you will need to enter in a NTP server. Some providers will provide you with one.

Change to the "System" tab



Primary NTP Server: au.pool.ntp.org

Change to the "Regional" tab



For NSW these settings seem to work:

Time Zone: GMT+11:00
Daylight Saving Time Rule: start=3/-1/7/3;end=10/-1/7/2;save=-1

Appendix G – Setting up a PSTN to VoIP Gateway

The Sipura SPA-3000 allows you to dial in from an external location, through the PSTN line and then dial out using a VoIP provider.

To do this you need to have a VoIP provider registered on the PSTN tab. The example below shows Astratel being registered.

Proxy and Registration			
Proxy:	sip03.astrasip.com.au	Use Outbound Proxy:	no
Outbound Proxy:		Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal

You also must enter your subscriber details, the same way as was done in the Line 1 registration.

Subscriber Information			
Display Name:	JMG Technology	User ID:	8888xxxx
Password:	*************	Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			

The next thing, that you need to is configure the PSTN to VoIP gateway using a PIN number.

PSTN-To-VoIP Gateway Setup			
PSTN-To-VoIP Gateway Enable:	yes	PSTN Caller Auth Method:	PIN
PSTN Ring Thru Line 1:	yes	PSTN PIN Max Retry:	3
PSTN CID For VoIP CID:	no	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:	
PSTN Caller ID Pattern:			
PSTN Access List:			
PSTN Caller 1 PIN:	1234	PSTN Caller 1 DP:	1
PSTN Caller 2 PIN:		PSTN Caller 2 DP:	1
PSTN Caller 3 PIN:		PSTN Caller 3 DP:	1
PSTN Caller 4 PIN:		PSTN Caller 4 DP:	1
PSTN Caller 5 PIN:		PSTN Caller 5 DP:	1
PSTN Caller 6 PIN:		PSTN Caller 6 DP:	1
PSTN Caller 7 PIN:		PSTN Caller 7 DP:	1
PSTN Caller 8 PIN:		PSTN Caller 8 DP:	1

PSTN Caller Auth Method: PIN
PSTN Caller 1 PIN: (whatever number you choose)

The VoIP call will be made with whatever VoIP provider you have registered on the PSTN Line tab.

At this stage you may want to change the time it takes for the SPA-3000 to pick up the PSTN line when you ring into it.

FXO Timer Values (sec)			
VoIP Answer Delay:	0	VoIP PIN Digit Timeout:	10
PSTN Answer Delay:	8	PSTN PIN Digit Timeout:	10
PSTN-To-VoIP Call Max Dur:	0	PSTN Ring Thru Delay:	1
VoIP-To-PSTN Call Max Dur:	0	PSTN Ring Thru CWT Delay:	3
VoIP DLG Refresh Intvl:	0	PSTN Ring Timeout:	5
PSTN Dialing Delay:	2	PSTN Dial Digit Len:	.1/.1
PSTN Hook Flash Len:	.25		

PSTN Answer Delay: 12; (Change this to whatever you think is good for you)
PSTN Dialing Delay: 2; (This seems to work well)

Appendix H – Proxy and Registration Settings For Common Providers

Astratel: Proxy and Registration Settings

Proxy and Registration			
Proxy:	sip03.astrasip.com.au	Use Outbound Proxy:	no
Outbound Proxy:		Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Voice Mail Server:			

Proxy: sip03.astrasip.com.au

MyNetFone: Proxy and Registration Settings

Proxy and Registration			
Proxy:	sip.myfone.com.au	Use Outbound Proxy:	yes
Outbound Proxy:	sip.myfone.com.au	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	240	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Voice Mail Server:			

Proxy: sip.myfone.com.au
Use Outbound Proxy: yes
Outbound Proxy: sip.myfone.com.au
Register Expires: 240

more to come .....

Appendix I – Sipbroker