

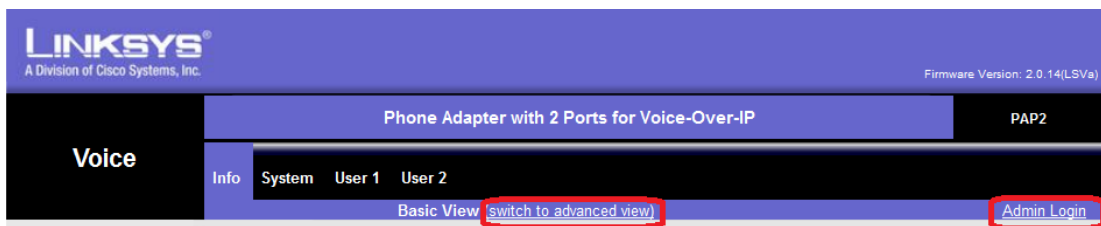
VoIP Setup Guide for Linksys PAP2

1. Connect a telephone handset to the **Phone 1** port



Check the IP Address of the phone adapter:

2. Press **** (i.e. press the star key four times)
3. Wait until you hear response “Linksys configuration menu—please enter the option followed by the # (hash) key or hang up to exit”
4. Enter 110#
5. The IP address will be read out - please note it down
6. Open your web browser (e.g. Internet Explorer, Mozilla Firefox, Google Chrome, etc)
7. Enter the **IP address** that you noted down in step 5 in the address bar
8. On the top right side of the window, click on **Admin Login**
9. Click on **(switch to advanced view)** to the left of the Admin Login



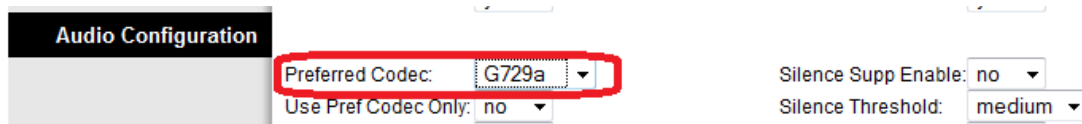
10. On the top menu, click on **Line 1**
11. For **NAT Mapping Enable** - select Yes
12. For **NAT Keep Alive Enable** - select Yes
13. For **Proxy** - enter sip00.mynetfone.com.au
14. For **Outbound Proxy** - enter sip00.mynetfone.com.au
15. For **Use Outbound Proxy** - select Yes
16. For **Register Expires** - enter 120

Streaming Audio Server (SAS)	Line Enable: <input type="text" value="yes"/>	SAS DLG Refresh Intvl: <input type="text" value="30"/>
NAT Settings	SAS Enable: <input type="text" value="no"/> SAS Inbound RTP Sink: <input type="text"/>	NAT Mapping Enable: <input type="text" value="yes"/> (circled) NAT Keep Alive Enable: <input type="text" value="yes"/> (circled) NAT Keep Alive Msg: <input type="text" value="\$NOTIFY"/> NAT Keep Alive Dest: <input type="text" value="\$PROXY"/>
Network Settings	SIP TOS/DiffServ Value: <input type="text" value="0x68"/> RTP TOS/DiffServ Value: <input type="text" value="0xb8"/>	Network Jitter Level: <input type="text" value="high"/>
SIP Settings	SIP Port: <input type="text" value="5060"/> EXT SIP Port: <input type="text"/> SIP Debug Option: <input type="text" value="none"/> Restrict Source IP: <input type="text" value="no"/>	SIP 100REL Enable: <input type="text" value="no"/> Auth Resync-Reboot: <input type="text" value="yes"/> RTP Log Intvl: <input type="text" value="0"/>
Call Feature Settings	Blind Attn-Xfer Enable: <input type="text" value="no"/> Xfer When Hangup Conf: <input type="text" value="yes"/>	MOH Server: <input type="text"/>
Proxy and Registration	Proxy: <input type="text" value="Mynetfone SIP Pro:"/> (circled) Outbound Proxy: <input type="text" value="Mynetfone SIP Pro:"/> (circled) Register: <input type="text" value="yes"/> Register Expires: <input type="text" value="120"/> (circled)	Use Outbound Proxy: <input type="text" value="yes"/> (circled) Use Out Proxy in Dialog: <input type="text" value="yes"/> Make Call Without Reg: <input type="text" value="no"/> Ans Call Without Reg: <input type="text" value="no"/>

17. For **Display Name** - enter your MyNetFone Number
18. For **User ID** - enter your MyNetFone Number
19. For **Password** - enter your MyNetFone Password

Subscriber Information	Display Name: <input type="text" value="Mynetfone Numbe"/> (circled) Password: <input type="text" value="Mynetfone Passwc"/> (circled) Auth ID: <input type="text"/> Mini Certificate: <input type="text"/> SRTP Private Key: <input type="text"/>	User ID: <input type="text" value="Mynetfone Number"/> (circled) Use Auth ID: <input type="text" value="no"/>
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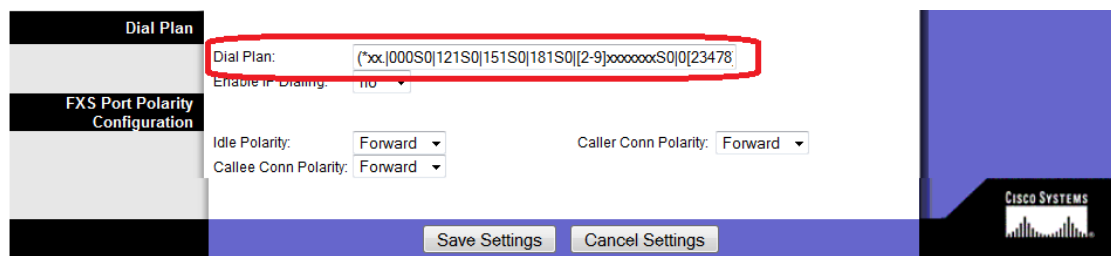
20. For **Preferred Codec** - select G729a



The screenshot shows the 'Audio Configuration' section of a web interface. The 'Preferred Codec' dropdown menu is highlighted with a red box and set to 'G729a'. Below it, the 'Use Pref Codec Only' dropdown is set to 'no'. To the right, 'Silence Supp Enable' is set to 'no' and 'Silence Threshold' is set to 'medium'.

21. For **Dial Plan** - enter the following:

(*xx.|000S0|121S0|151S0|181S0|[2-9]xxxxxxS0|0[23478]xxxxxxS0|0011xxx.|1800xxxxxxS0|1300xxxxxxS0|13[1-9]xxxS0|xxx.)



The screenshot shows the 'Dial Plan' configuration section. The 'Dial Plan' text field is highlighted with a red box and contains the string: (*xx.|000S0|121S0|151S0|181S0|[2-9]xxxxxxS0|0[23478]. Below the text field, there are several dropdown menus for 'Enable IP Dialing', 'Idle Polarity', 'Callee Conn Polarity', and 'Caller Conn Polarity', all of which are set to 'Forward'. At the bottom of the section, there are 'Save Settings' and 'Cancel Settings' buttons. The Cisco Systems logo is visible in the bottom right corner.

To setup Phone port 2 - click on Line 2 and follow the steps from 11- 21.

22. Click on **Save Settings**