

Linksys:

Connect Linksys and login in it using IE (internet explorer).

<http://x.x.x.x/>

For Line 1

1. Click on line 1 and enter the switch port and switch IP in the text box SIP port and Proxy text box respectively.
2. Put a user ID/test pin in the text boxes Display Name, User ID and Auth ID
3. Put the corresponding password of that pin in the field Password.

Info	System	SIP	Regional	Line 1	Line 2	User 1	User 2
Basic View (switch to advanced view)							
Line Enable:	yes						
SIP Port:	5060						
Proxy:	202.4.97.11			Register:	yes		
Make Call Without Reg:	no			Register Expires:	3600		
Ans Call Without Reg:	no						
Display Name:	09611200078			User ID:	09611200078		
Password:	*****			Use Auth ID:	no		
Auth ID:	09611200078						
Call Waiting Serv:	no			Block CID Serv:	yes		
Block ANC Serv:	yes			Dist Ring Serv:	yes		
Cfwd All Serv:	yes			Cfwd Busy Serv:	yes		

Audio Configuration	
Preferred Codec:	G729a
Use Pref Codec Only:	yes
DTMF Tx Method:	Auto
Silence Supp Enable:	yes
FAX CED Detect Enable:	yes

4. In the audio configuration, Select preferred codec G729a and use Pref Codec only yes from the dropdown list.
5. And finally click Save Settings.

For Line 2

Audio Configuration	
Preferred Codec:	G729a
Use Pref Codec Only:	no
DTMF Tx Method:	Auto
Silence Supp Enable:	no
FAX CED Detect Enable:	yes

Save Settings Cancel Settings

1. Select entry as like line 1 except use pref codec is no on line 2.
2. And finally click Save Settings

After configured line 1 and 2, click Switch to Advanced View and then SIP. Change the value RTP packet size to 0.020

Voice								
Info	System	SIP	Provisioning	Regional	Line 1	Line 2	User 1	User 2
Advanced View (switch to basic view)								
SIP Parameters								
Max Forward:	70	Max Redirection:	5					
Max Auth:	2	SIP User Agent Name:	\$VERSION					
SIP Server Name:	\$VERSION	SIP Reg User Agent Name:						
SIP Accept Language:		DTMF Relay MIME Type:	application/dtmf-r					
Hook Flash MIME Type:	application/hook-	Remove Last Reg:	no					
Use Compact Header:	no	Escape Display Name:	no					
RFC 2543 Call Hold:	yes	Softswitch Features:						
SIP Timer Values (sec)								

Need to change value here.

RTP Parameters			
RTP Port Min:	16384	RTP Port Max:	16482
RTP Packet Size:	0.030	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no
Stats In BYE:	no		