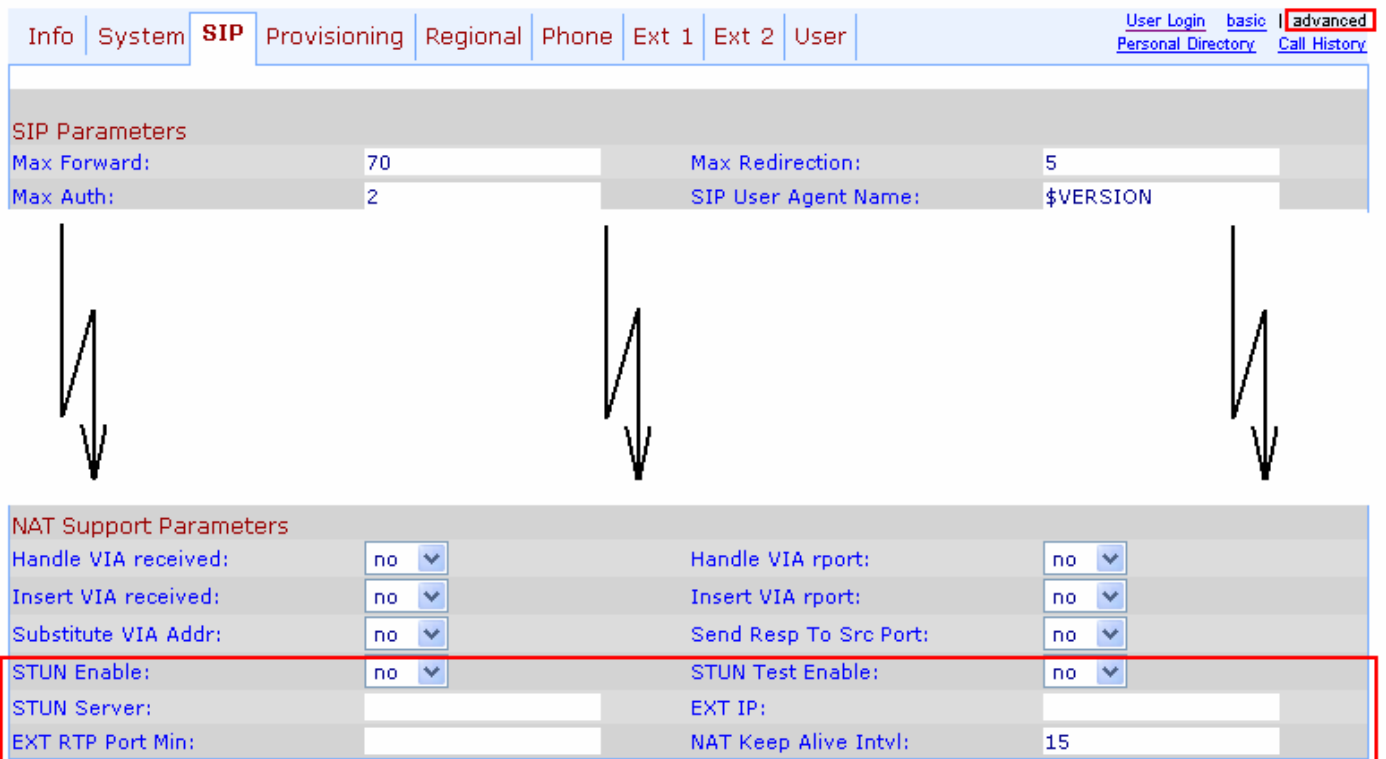


1. Open your web browser and navigate to the URL;

http://<Linksys PAP IP>/admin/advanced

2. Go to "SIP" Page and check the configuration to match the below highlighted area;



Info System **SIP** Provisioning Regional Phone Ext 1 Ext 2 User [User Login](#) [basic](#) **advanced**
[Personal Directory](#) [Call History](#)

SIP Parameters

| | | | |
|--------------|----|----------------------|-----------|
| Max Forward: | 70 | Max Redirection: | 5 |
| Max Auth: | 2 | SIP User Agent Name: | \$VERSION |

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NAT Support Parameters

| | | | |
|----------------------|----|------------------------|----|
| Handle VIA received: | no | Handle VIA rport: | no |
| Insert VIA received: | no | Insert VIA rport: | no |
| Substitute VIA Addr: | no | Send Resp To Src Port: | no |
| STUN Enable: | no | STUN Test Enable: | no |
| STUN Server: | | EXT IP: | |
| EXT RTP Port Min: | | NAT Keep Alive Intvl: | 15 |

3. Go to "Ext 1", and configure it to look as displayed below:

| | | | | | | | | | | | |
|------|--------|-----|--------------|----------|-------|--------------|-------|------|------------------------------------|------------------------------|--------------------------|
| Info | System | SIP | Provisioning | Regional | Phone | Ext 1 | Ext 2 | User | User Login | basic | advanced |
| | | | | | | | | | Personal Directory | Call History | |

General

Line Enable:

Share Line Appearance

Share Ext: Shared User ID:

Subscription Expires:

NAT Settings

NAT Mapping Enable: NAT Keep Alive Enable:

NAT Keep Alive Msg: NAT Keep Alive Dest:

Network Settings

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Proxy and Registration

Proxy: Use Outbound Proxy:

Outbound Proxy: Use OB Proxy In Dialog:

Register: Make Call Without Reg:

Register Expires: Ans Call Without Reg:

Use DNS SRV: DNS SRV Auto Prefix:

Proxy Fallback Intvl: Proxy Redundancy Method:

Subscriber Information

Display Name: User ID:

Password: Use Auth ID:

Auth ID:

Mini Certificate:

SRTP Private Key:

Audio Configuration

Preferred Codec: Use Pref Codec Only:

Use Pref Codec Only:

G726-16 Enable: G726-24 Enable:

G726-32 Enable: G726-40 Enable:

Release Unused Codec: DTMF Process AVT:

Silence Supp Enable: DTMF Tx Method:

Dial Plan

Dial Plan: Enable IP Dialing:

Ref A:

Proxy: sip.checkcdr.com

Outbound Proxy: obproxy.checkcdr.com:7899

Note: Using Proxy as `sip.checkcdr.com` would use default sip port 5060.
For Customers who have port blocking problem with their ISP, could try the following variations as Proxy:

`sip.checkcdr.com:3060`
`sip.checkcdr.com:1175`

Ref B:

Preferred codecs are **G723** and **G729a**

Use preferred codec only: **YES**

Note: If you're using PAP2T, then you're able to set both lines on G729a. However incase of PAP2, this cannot be done due to a limitation of the device itself.

Ref C:

Dial Plan:

`(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxxx.)`

4. Configure "Ext 2" as well to with same configuration. The Only difference would be SIP PORT: should be something different than Ext 1,

Ext 1 SIP Port Default: 5060

Ext 2 SIP Port Default: 5061

5. Now press "Save Settings" button. Device would restart and you'll be ready to call!

If you experience any problem with configurations, please do feel free to call for support, and we'll be very glad to assist you.