



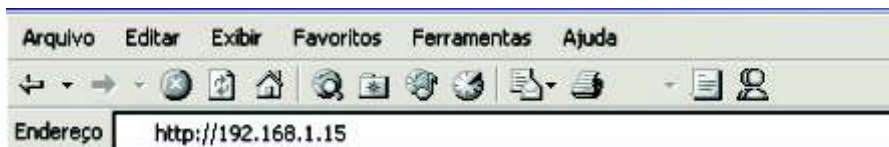
Linksys PAP2

Para acessar as configurações do equipamento, você deverá pegar o endereço IP, discando pelo telefone (plugado no Phone 1) 4 vezes * e 110#. Ao pegar o endereço IP, digite-o no seu navegador para acessar o menu de configurações do seu ata.

Por exemplo:

Endereço IP adquirido na função **** e 110# : 192.168.1.15

Digite no seu navegador o endereço <http://192.168.1.15>, conforme a figura abaixo:



Para Configurar o seu PAP2, é necessário que você acesse o menu de seu ATA, clique em **Admin Login** (situado no canto superior a direita) e **Switch to Advanced View** (situado abaixo de Line 1) e siga as configurações abaixo:



- Clique em System e configure os parâmetros de REDE.
OBS: lembrando que o ATA PAP2 não configura o parâmetro PPPoE.

LINKSYS
A Division of Cisco Systems, Inc. Firmware Version: 2.1.9(15c)

Voice Phone Adapter with 2 Ports for Voice Over-IP **PAP2**

Info **System** SIP Provisioning Regional Line 1 Line 2 User 1 User 2

[Advanced View](#) [\(switch to basic view\)](#) [User Login](#)

System Configuration

Restricted Access Domains:

Enable Web Server: yes no Web Server Port:

Enable Web Admin Access: yes no Admin Passwd:

User Password:

Internet Connection Type

DHCP: yes no NetMask:

Static IP:

Gateway:

Optional Network Configuration

HostName:

Primary DNS:

DNS Server Order: Manual Domain:

Syslog Server:

Debug Level: 0 Secondary DNS:

Secondary NTP Server:

DNS Query Mode: Parallel

Debug Server:

Primary NTP Server:



- Clique em **Provisioning**, altere a opção **circulada em vermelho** e logo em seguida clique em **Save Settings**;

Voice Info System SIP **Provisioning** Regional Line 1 Line 2 User 1 User 2

Advanced View [\(switch to basic view\)](#) [User Login](#)

Configuration Profile

Provision Enable:	no	Resync On Reset:	yes
Resync Random Delay:	2	Resync Periodic:	3600
Resync Error Retry Delay:	3600	Forced Resync Delay:	14400
Resync From SIP:	yes	Resync After Upgrade Attempt:	yes
Resync Trigger 1:			
Resync Trigger 2:			
Resync Fails On FNF:	no		
Profile Rule:	/int.cfg		
Profile Rule B:			
Profile Rule C:			
Profile Rule D:			
Log Resync Request Msg:	\$PN \$MAC -- Requesting resync \$SCHEME://\$SERVIP:		
Log Resync Success Msg:	\$PN \$MAC -- Successful resync \$SCHEME://\$SERVIP:		
Log Resync Failure Msg:	\$PN \$MAC -- Resync failed: \$ERR		
Report Rule:			

Save Settings **Cancel Settings**

CISCO SYSTEMS



- Ao voltar a tela, clique em **Line 1** altere as opções **circuladas em vermelho** e logo em seguida clique em **Save Settings**;

Voice		Info	System	SIP	Provisioning	Regional	Line 1	Line 2	User 1	User 2
Advanced View (switch to basic view) User Login										
Streaming Audio Server (SAS)	Line Enable:	<input type="text" value="yes"/>								
	SAS Enable:	<input type="text" value="no"/>								
	SAS Inbound RTP Sink:	<input type="text"/>								
	SAS DLG Refresh Intvl:	<input type="text" value="30"/>								
NAT Settings	NAT Mapping Enable:	<input type="text" value="no"/>								
	NAT Keep Alive Msg:	<input type="text" value="\$NOTIFY"/>								
	NAT Keep Alive Enable:	<input type="text" value="no"/>								
	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"/>								
	Jitter Buffer Adjustment:	<input type="text" value="up and down"/>								
SIP Settings	SIP Port:	<input type="text" value="5060"/>								
	EXT SIP Port:	<input type="text"/>								
	SIP Proxy-Require:	<input type="text"/>								
	SIP GUID:	<input type="text" value="no"/>								
	RTP Log Intvl:	<input type="text" value="0"/>								
	Referor Bye Delay:	<input type="text" value="4"/>								
	Referee Bye Delay:	<input type="text" value="0"/>								
	Sticky 183:	<input type="text" value="no"/>								
	SIP 100REL Enable:	<input type="text" value="no"/>								
	Auth Resync-Reboot:	<input type="text" value="yes"/>								
	SIP Remote-Party-ID:	<input type="text" value="no"/>								
	SIP Debug Option:	<input type="text" value="none"/>								
	Restrict Source IP:	<input type="text" value="no"/>								
	Refer Target Bye Delay:	<input type="text" value="0"/>								
	Refer-To Target Contact:	<input type="text" value="no"/>								
Call Feature Settings	Blind Attn-Xfer Enable:	<input type="text" value="no"/>								
	Xfer When Hangup Conf:	<input type="text" value="yes"/>								
	Conference Bridge Ports:	<input type="text" value="3"/>								
	MOH Server:	<input type="text"/>								
	Conference Bridge URL:	<input type="text"/>								
Proxy and Registration	Proxy:	<input type="text" value="189.114.228.20"/>								
	Outbound Proxy:	<input type="text" value="189.114.228.20"/>								
	Register:	<input type="text" value="yes"/>								
	Register Expires:	<input type="text" value="3600"/>								
	Use DNS SRV:	<input type="text" value="yes"/>								
	Proxy Fallback Intvl:	<input type="text" value="3600"/>								
	Voice Mail Server:	<input type="text"/>								
	Use Outbound Proxy:	<input type="text" value="yes"/>								
	Use OB 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"/>								
	DNS SRV Auto Prefix:	<input type="text" value="no"/>								
	Proxy Redundancy Method:	<input type="text" value="Normal"/>								
Subscriber Information	Display Name:	<input type="text" value="Seu nome"/>								
	Password:	<input type="text" value="Sua senha"/>								
	User ID:	<input type="text" value="Seu login"/>								
	Use Auth ID:	<input type="text" value="no"/>								
	Auth ID:	<input type="text"/>								
	Mini Certificate:	<input type="text"/>								
	SRTP Private Key:	<input type="text"/>								



Supplementary Service Subscription	Call Waiting Serv.: yes	Block CID Serv.: yes
	Block ANC Serv.: yes	Dist Ring Serv.: yes
	Cfwd All Serv.: yes	Cfwd Busy Serv.: yes
	Cfwd No Ans Serv.: yes	Cfwd Sel Serv.: yes
	Cfwd Last Serv.: yes	Block Last Serv.: yes
	Accept Last Serv.: yes	DND Serv.: yes
	CID Serv.: yes	OWCID Serv.: yes
	Call Return Serv.: yes	Call Back Serv.: yes
	Three Way Call Serv.: yes	Three Way Conf Serv.: yes
	Attn Transfer Serv.: yes	Unathn Transfer Serv.: yes
	MVM Serv.: yes	VVM Serv.: yes
	Speed Dial Serv.: yes	Secure Call Serv.: yes
	Referral Serv.: yes	Feature Dial Serv.: yes
	Service Announcement Serv.: no	
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
	FAX Codec Symmetric: 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	
Dial Plan	Dial Plan: [*]xx[*]x[3469]11000[2-9]xxxxxx[1]xx[2-9]xxxxxx50xxxx	
	Enable IP Dialing: no	Emergency Number:
FXS Port Polarity Configuration	Idle Polarity: Forward	Caller Conn Polarity: Forward
	Callee Conn Polarity: Forward	
	Save Settings	Cancel Settings

Assim o seu ATA estará configurado e pronto para efetuar chamadas VoIP. Caso haja alguma irregularidade, recomendamos que você verifique os campos preenchidos com atenção.