



## Tutorial SPA3102 e PAP2T

Olá pessoal! Nesse tutorial iremos entroncar dois atas chamados de SPA 3102 e PAP2T. Caso ainda não tenha o [SPA3102](#) clique aqui, ou o [PAP2T](#) clique aqui.

Antes de começarmos sugiro que dê um reset nos dois aparelhos para voltar as configurações “default”. Certamente os dois já estão devidamente conectados e ligados. Suponho que você saiba o básico de redes e protocolos IP.

Obs:

RESETANDO: Digite no telefone \*\*\*\* 73738 # 1#

IP DHCP – para ouvir digite \*\*\*\* 110 #

Configurando o SPA3102

-Primeiro acesso:

Para fazer o primeiro acesso ao aparelho você precisa conectar o cabo de rede na porta ETHERNET (LAN). Após isso, coloque um apelido de IP no seu computador no mesmo “range” (192.168.0.X), e digite no browser o seguinte IP “192.168.0.1”.

Na interface web clique em “Admin Login” e “advanced”, respectivamente.



The screenshot shows the Linksys Phone Adapter Configuration web interface. At the top, it displays the Linksys logo and 'A Division of Cisco Systems, Inc.' on the left, and 'Linksys Phone Adapter Configuration' on the right. Below this is a navigation bar with 'Router' and 'Voice' tabs. The main content area has 'Status' and 'Wan Setup' tabs. In the top right corner, there are three links: 'Admin Login', 'basic', and 'advanced', with 'Admin Login' and 'advanced' circled in red. The 'Product Information' section includes fields for Product Name (SPA-3102), Serial Number (FM600J015776), Software Version (5.2.13(GW002)), Hardware Version (1.4.5), MAC Address (00E08A2259A), Client Certificate (Installed), and Customization (Open). The 'System Status' section includes Current Time (1/1/2003 12:00:33), Elapsed Time (00:00:33), Wan Connection Type (DHCP), Current IP (0.0.0.0), Host Name (SipuraSPA), Domain, Current Netmask (0.0.0.0), Current Gateway (0.0.0.0), Primary DNS, Secondary DNS, LAN IP Address (192.168.0.1), Broadcast Pkts Sent (5), Broadcast Bytes Sent (1710), Broadcast Pkts Recv (191), Broadcast Bytes Recv (17147), Broadcast Pkts Dropped (0), Broadcast Bytes Dropped (0), and WAN Link Status. At the bottom of the configuration area are two buttons: 'Undo All Changes' and 'Submit All Changes'. Below the buttons are links for 'Admin Login', 'basic', and 'advanced'.

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-Configurando porta WAN:

>ROUTER > WAN SETUP

- Internet Connection Settings

Connection Type: Static IP

- Static IP Settings

Static IP: Coloque um IP fixo.

NetMask: Coloque a máscara da rede.

Gateway: Coloque o IP do PAP2T.

## - Remote Management

Enable WAN Web Server: yes.

Agora você clique em “Submit All Chenges” para salvar as alterações. A parte de agora coloque o cabo de rede na portaINTERNET(WAN), e acesse o gateway com o IP que foi colocado.



The screenshot shows the Linksys Phone Adapter Configuration interface. The 'Router' tab is selected, and the 'Wan Setup' sub-tab is active. The 'Internet Connection Settings' section is expanded, showing 'Connection Type' set to 'DHCP'. The 'Static IP Settings' section has 'Static IP' and 'NetMask' fields circled in red. The 'Remote Management' section has 'Enable WAN Web Server' set to 'no', also circled in red. Other sections include 'PPPoE Settings', 'Optional Settings', 'MAC Clone Settings', and 'QoS Settings'.

>VOICE -> PSTN LINE

## - SIP Settings

SIP Port: 5061

## - Proxy and Registration

Proxy: Coloque o “(IP do PAP2T):5060”. Ex: 192.168.0.2:5060.

Make Call Without Reg: yes.

Ans Call Without Reg: yes.

## - Subscriber Information

User ID: Coloque a SIP que ligará o SPA 3102 com PAP2T.

Use Auth ID: no.

Network Settings			
SIP ToS/DiffServ Value:	0x68	SIP CoS Value:	3 [0-7]
RTP ToS/DiffServ Value:	0xb8	RTP CoS Value:	6 [0-7]
Network Jitter Level:	high	Jitter Buffer Adjustment:	up and down
SIP Settings			
SIP Transport:	UDP	SIP Port:	5061
SIP 100REL Enable:	no	EXT SIP Port:	
Auth Resync-Reboot:	yes	SIP Proxy-Require:	
SIP Remote-Party-ID:	yes	SIP GUID:	no
SIP Debug Option:	none	RTP Log Intvl:	0
Restrict Source IP:	no	Referor Bye Delay:	4
Refer Target Bye Delay:	0	Referee Bye Delay:	0
Refer-To Target Contact:	no	Sticky 183:	no
Auth INVITE:	no	Use Anonymous With RPID:	yes
Use Local Addr In FROM:	no		
Proxy and Registration			
Proxy:			
Outbound Proxy:			
Use Outbound Proxy:	no	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
Subscriber Information			
Display Name:		User ID:	
Password:		Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			
Audio Configuration			
Preferred Codec:	G711u	Silence Supp Enable:	no
Use Pref Codec Only:	no	Echo Canc Enable:	yes
G729a Enable:	yes	Echo Canc Adapt Enable:	yes
G723 Enable:	yes	Echo Supp Enable:	yes
G726-16 Enable:	yes	FAX CED Detect Enable:	yes
G726-32 Enable:	yes	FAX CNG Detect Enable:	yes

## - Dial Plans

Dial Plan 2: Coloque “(<S0:User ID@IP DO PAP2T:5060>)”.



<b>Audio Configuration</b>			
Preferred Codec:	G711u	Silence Supp Enable:	no
Use Pref Codec Only:	no	Echo Canc Enable:	yes
G729a Enable:	yes	Echo Canc Adapt Enable:	yes
G723 Enable:	yes	Echo Supp Enable:	yes
G726-16 Enable:	yes	FAX CED Detect Enable:	yes
G726-24 Enable:	yes	FAX CNG Detect Enable:	yes
G726-32 Enable:	yes	FAX Passthru Codec:	G711u
G726-40 Enable:	yes	FAX Codec Symmetric:	yes
DTMF Process INFO:	yes	FAX Passthru Method:	NSE
DTMF Process AVT:	yes	DTMF Tx Method:	Auto
DTMF Tx Mode:	Strict	DTMF Tx Strict Hold Off Time:	40
Release Unused Codec:	yes	FAX Process NSE:	yes
Symmetric RTP:	yes	FAX Disable ECAN:	no
Audio Dump Option1:	none	Audio Dump Option2:	none
<b>Dial Plans</b>			
Dial Plan 1:	(xx.)		
Dial Plan 2:	(xx.)		
Dial Plan 3:	(xx.)		
Dial Plan 4:	(xx.)		
Dial Plan 5:	(xx.)		
Dial Plan 6:	(xx.)		
Dial Plan 7:	(xx.)		
Dial Plan 8:	(xx.)		
<b>VoIP-To-PSTN Gateway Setup</b>			
VoIP-To-PSTN Gateway Enable:	yes	VoIP Caller Auth Method:	none
VoIP PIN Max Retry:	3	One Stage Dialing:	yes
Line 1 VoIP Caller DP:	1	VoIP Caller Default DP:	1
Line 1 Fallback DP:	none		
VoIP Caller ID Pattern:			
VoIP Access List:			
VoIP Caller 1 PIN:		VoIP Caller 1 DP:	1
VoIP Caller 2 PIN:		VoIP Caller 2 DP:	1
VoIP Caller 3 PIN:		VoIP Caller 3 DP:	1
VoIP Caller 4 PIN:		VoIP Caller 4 DP:	1
VoIP Caller 5 PIN:		VoIP Caller 5 DP:	1
VoIP Caller 6 PIN:		VoIP Caller 6 DP:	1

## - PSTN-To-Voip Gateway Setup

**PSTN Caller Default DP: 2**

**PSTN Answer Delay: 0**

**VoIP PIN Digit Timeout: 5**

**PSTN PIN Digit Timeout: 5**

VoIP User 6 Auth ID:		VoIP User 6 DP:	1
VoIP User 6 Password:			
VoIP User 7 Auth ID:		VoIP User 7 DP:	1
VoIP User 7 Password:			
VoIP User 8 Auth ID:		VoIP User 8 DP:	1
VoIP User 8 Password:			
<b>PSTN-To-VoIP Gateway Setup</b>			
PSTN-To-VoIP Gateway Enable:	yes	PSTN Caller Auth Method:	none
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:		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
<b>FXO Timer Values (sec)</b>			
VoIP Answer Delay:	0	VoIP PIN Digit Timeout:	10
PSTN Answer Delay:	16	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:	1	PSTN Dial Digit Len:	.1/.1
PSTN Hook Flash Len:	.25		
<b>PSTN Disconnect Detection</b>			
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:	480@-30,620@-30:4(.25/.25/1+2)		

Após esse procedimentos salve as alterações. Agora vamos aponta o PAP2T para o SPA 3102.

## CONFIGURANDO O PAP2T

### -Primeiro acesso:

Para fazer o primeiro acesso ao aparelho, você precisa conectar em das porta fxs um telefone analógico, e digitar os código citados no início do documento. A ordem do procedimento é; primeiro o reset (\*\*\*\* 73738 # 1#) depois o aparelho emitira um IP DHCP automático (código para ouvir o IP:\*\*\*\* 110 #).

Após acessar clique em “Admin Login” e “advanced”, respectivamente. Como mostrado na imagem abaixo.

**LINKSYS**  
A Division of Cisco Systems, Inc. Firmware Version: 5.1.6(LS)

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

Info System User 1 User 2

[Basic View \(switch to advanced view\)](#) [Admin Login](#)

System Information	DHCP: <b>Enabled</b> Host Name: LinksysPAP Current Netmask: 255.255.255.0 Primary DNS: 8.8.8.8 Secondary DNS:	Current IP: 20.20.20.202 Domain: sms Current Gateway: 20.20.20.1
Product Information	Product Name: PAP2T Software Version: 5.1.6(LS) MAC Address: 0021291E6873 Customization: Open	Serial Number: FL100J0F6874 Hardware Version: 5.2.1 Client Certificate: Installed
System Status	Current Time: 1/1/2003 12:01:54 Broadcast Pkts Sent: 9 Broadcast Pkts Recv: 786 Broadcast Pkts Dropped: 0 RTP Packets Sent: 0 RTP Packets Recv: 0 SIP Messages Sent: 0 SIP Messages Recv: 0 External IP:	Elapsed Time: 00:01:54 Broadcast Bytes Sent: 3078 Broadcast Bytes Recv: 57721 Broadcast Bytes Dropped: 0 RTP Bytes Sent: 0 RTP Bytes Recv: 0 SIP Bytes Sent: 0 SIP Bytes Recv: 0
Line 1 Status	Display Name: Hook State: <b>On</b> Last Registration At: Message Waiting: <b>No</b> Last Called Number: Mapped SIP Port: Call 1 State: <b>Idle</b>	User ID: Registration State: <b>Offline</b> Next Registration In: Call Back Active: <b>No</b> Last Caller Number: Call 2 State: <b>Idle</b>

>System > Internet Connection Type

DHCP: no

Static IP: Coloque um IP fixo.

NetMask: Coloque a máscara da rede.

Gateway: Coloque o IP do SPA 3102.

**LINKSYS**  
A Division of Cisco Systems, Inc. Firmware Version: 5.1.6(LS)

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

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

Basic View [\(switch to advanced view\)](#) [User Login](#)

**System Configuration**

Enable Web Server:  User Password:

**Internet Connection Type**

DHCP:  Static IP:  NetMask:

Gateway:

**Optional Network Configuration**

HostName:  Domain:

Primary DNS:  Secondary DNS:

DNS Query Mode:  Syslog Server:

Debug Server:  Debug Level:

>Line 1 > Proxy and Registration

Proxy: Coloque o "(IP do SPA 3102):5061". Ex: 192.168.0.1:5061

Make Call Without Reg: yes

Ans Call Without Reg: yes

>Subscriber Information

User ID: Coloque a SIP que ligará o PAP2T com SPA 3102.

Use Auth ID: no



Blind Attn-Xfer Enable:	<input type="text" value="no"/>	MOH Server:	<input type="text"/>
Xfer VWhen Hangup Conf:	<input type="text" value="yes"/>	Conference Bridge URL:	<input type="text"/>
Conference Bridge Ports:	<input type="text" value="3"/>		
<b>Proxy and Registration</b>			
Proxy:	<input type="text"/>	Use Outbound Proxy:	<input type="text" value="no"/>
Outbound Proxy:	<input type="text"/>	Use OB Proxy In Dialog:	<input type="text" value="yes"/>
Register:	<input type="text" value="yes"/>	Make Call Without Reg:	<input type="text" value="no"/>
Register Expires:	<input type="text" value="3600"/>	Ans Call Without Reg:	<input type="text" value="no"/>
Use DNS SRV:	<input type="text" value="no"/>	DNS SRV Auto Prefix:	<input type="text" value="no"/>
Proxy fallback Intvt:	<input type="text" value="3600"/>	Proxy Redundancy Method:	<input type="text" value="Normal"/>
Voice Mail Server:	<input type="text"/>	Mailbox Subscribe Expires:	<input type="text" value="2147483647"/>
<b>Subscriber Information</b>			
Display Name:	<input type="text"/>	User ID:	<input type="text"/>
Password:	<input type="text"/>	Use Auth ID:	<input type="text" value="no"/>
Auth ID:	<input type="text"/>		
Mini Certificate:	<input type="text"/>		
SRTP Private Key:	<input type="text"/>		
<b>Supplementary Service Subscription</b>			
Call Waiting Serv:	<input type="text" value="yes"/>	Block CID Serv:	<input type="text" value="yes"/>
Block ANC Serv:	<input type="text" value="yes"/>	Dist Ring Serv:	<input type="text" value="yes"/>
Cfwd All Serv:	<input type="text" value="yes"/>	Cfwd Busy Serv:	<input type="text" value="yes"/>
Cfwd No Ans Serv:	<input type="text" value="yes"/>	Cfwd Sel Serv:	<input type="text" value="yes"/>
Cfwd Last Serv:	<input type="text" value="yes"/>	Block Last Serv:	<input type="text" value="yes"/>
Accept Last Serv:	<input type="text" value="yes"/>	DND Serv:	<input type="text" value="yes"/>
CID Serv:	<input type="text" value="yes"/>	CWCID Serv:	<input type="text" value="yes"/>
Call Return Serv:	<input type="text" value="yes"/>	Call Back Serv:	<input type="text" value="yes"/>
Three Way Call Serv:	<input type="text" value="yes"/>	Three Way Conf Serv:	<input type="text" value="yes"/>
Attn Transfer Serv:	<input type="text" value="yes"/>	Unattn Transfer Serv:	<input type="text" value="yes"/>
M/M Serv:	<input type="text" value="yes"/>	V/M/M Serv:	<input type="text" value="yes"/>

## >Audio Configuration

Release unused Codec: yes

## >Dial Plan

Dial Plan: Apague tudo e coloque “(<S0:User ID@IP DO SPA 3102:5061>)”.

**PRONTO!!! Seus atas estão configurado e funcionando!**