

Linksys SPA3102

LINKSYS[®]
A Division of Cisco Systems, Inc.
Linksys Phone Adapter Configuration

Router
Voice

Info
System
SIP
Regional
Line 1
PSTN Line
User 1
PSTN User

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basic
advanced

Response Status Code Handling

SIT1 RSC:	<input type="text"/>	SIT2 RSC:	<input type="text"/>
SIT3 RSC:	<input type="text"/>	SIT4 RSC:	<input type="text"/>

RTP Parameters

RTP Port Min:	16384	RTP Port Max:	16482
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SDP Payload Types

NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
G726r16 Dynamic Payload:	98	G726r24 Dynamic Payload:	97
G726r40 Dynamic Payload:	96	G729b Dynamic Payload:	99

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Line Enable:

SIP Settings

SIP Port:

Proxy and Registration

Proxy:

Register: <input type="text" value="yes"/>	Make Call Without Reg: <input type="text" value="no"/>
Register Expires: <input type="text" value="300"/>	Ans Call Without Reg: <input type="text" value="no"/>

Subscriber Information

Display Name: <input type="text" value="028881234"/>	User ID: <input type="text" value="028881234"/>
Password: <input type="text" value="*****"/>	Use Auth ID: <input type="text" value="yes"/>
Auth ID: <input type="text" value="02888 1234"/>	

Supplementary Service Subscription

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 Redial 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"/>	MWI Serv: <input type="text" value="yes"/>
VMWI Serv: <input type="text" value="yes"/>	

Audio Configuration

Audio Configuration

Preferred Codec: Silence Supp Enable:
Use Pref Codec Only: FAX CED Detect Enable:
DTMF Tx Method:

Gateway Accounts

VoIP Fallback To PSTN

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Linksys Phone Adapter Configuration

Router | **Voice** | Info | System | SIP | Regional | Line 1 | PSTN Line | **User 1** | PSTN User | [User Login](#) | [basic](#) | [advanced](#)

Call Forward Settings

Cfwd All Dest: Cfwd Busy Dest:
Cfwd No Ans Dest: Cfwd No Ans Delay:

Selective Call Forward Settings

Cfwd Sel1 Caller: <input type="text"/>	Cfwd Sel1 Dest: <input type="text"/>
Cfwd Sel2 Caller: <input type="text"/>	Cfwd Sel2 Dest: <input type="text"/>
Cfwd Sel3 Caller: <input type="text"/>	Cfwd Sel3 Dest: <input type="text"/>
Cfwd Sel4 Caller: <input type="text"/>	Cfwd Sel4 Dest: <input type="text"/>
Cfwd Sel5 Caller: <input type="text"/>	Cfwd Sel5 Dest: <input type="text"/>
Cfwd Sel6 Caller: <input type="text"/>	Cfwd Sel6 Dest: <input type="text"/>
Cfwd Sel7 Caller: <input type="text"/>	Cfwd Sel7 Dest: <input type="text"/>
Cfwd Sel8 Caller: <input type="text"/>	Cfwd Sel8 Dest: <input type="text"/>
Cfwd Last Caller: <input type="text"/>	Cfwd Last Dest: <input type="text"/>
Block Last Caller: <input type="text"/>	Accept Last Caller: <input type="text"/>

Supplementary Service Settings

CW Setting: <input type="text" value="yes"/>	Block CID Setting: <input type="text" value="no"/>
Block ANC Setting: <input type="text" value="no"/>	DND Setting: <input type="text" value="no"/>
CID Setting: <input type="text" value="yes"/>	CWCID Setting: <input type="text" value="yes"/>
Dist Ring Setting: <input type="text" value="yes"/>	Secure Call Setting: <input type="text" value="no"/>
Message Waiting: <input type="text" value="no"/>	Accept Media Loopback Request: <input type="text" value="automatic"/>
Media Loopback Mode: <input type="text" value="source"/>	Media Loopback Type: <input type="text" value="media"/>

Router		Voice					
Info	System	SIP	Regional	Line 1	PSTN Line	User 1	PSTN User
							User Login basic advanced
Product Information							
Product Name:	SPA-3102			Serial Number:	FM600F617997		
Software Version:	5.1.5(GWa)			Hardware Version:	1.1.5		
MAC Address:	000E08CCBAF8			Client Certificate:	Installed		
Customization:	Open						
System Status							
Current Time:	6/29/2007 15:51:58			Elapsed Time:	01:05:57		
RTP Packets Sent:	2585			RTP Bytes Sent:	620400		
RTP Packets Recv:	3876			RTP Bytes Recv:	620160		
SIP Messages Sent:	9			SIP Bytes Sent:	5960		
SIP Messages Recv:	8			SIP Bytes Recv:	5271		
External IP:							
Line 1 Status							
Hook State:	On			Registration State:	Registered		
Last Registration At:	6/29/2007 15:45:41			Next Registration In:	3193 s		
Message Waiting:	No			Call Back Active:	No		
Last Called Number:				Last Caller Number:	02 [REDACTED]		
Mapped SIP Port:	Yahoo!:						
Call 1 State:	Idle			Call 2 State:	Idle		
Call 1 Tone:	None			Call 2 Tone:	None		
Call 1 Encoder:				Call 2 Encoder:			
Call 1 Decoder:				Call 2 Decoder:			
Call 1 FAX:				Call 2 FAX:			
Call 1 Type:				Call 2 Type:			
Call 1 Remote Hold:				Call 2 Remote Hold:			
Call 1 Callback:				Call 2 Callback:			
Call 1 Peer Name:				Call 2 Peer Name:			
Call 1 Peer Phone:				Call 2 Peer Phone:			
Call 1 Duration:				Call 2 Duration:			
Call 1 Packets Sent:				Call 2 Packets Sent:			
Call 1 Packets Recv:				Call 2 Packets Recv:			
Call 1 Bytes Sent:				Call 2 Bytes Sent:			
Call 1 Bytes Recv:				Call 2 Bytes Recv:			
Call 1 Decode Latency:				Call 2 Decode Latency:			
Call 1 Jitter:				Call 2 Jitter:			
Call 1 Round Trip Delay:				Call 2 Round Trip Delay:			
Call 1 Round Trip Delay:				Call 2 Round Trip Delay:			
Call 1 Packets Lost:				Call 2 Packets Lost:			
Call 1 Packet Error:				Call 2 Packet Error:			
Call 1 Mapped RTP Port:				Call 2 Mapped RTP Port:			
Call 1 Media Loopback:				Call 2 Media Loopback:			
PSTN Line Status							
Hook State:	On			Line Voltage:	52 (V)		
Loop Current:	0.0 (mA)			Registration State:	Not Registered		
Last Registration At:				Next Registration In:			
Last Called VoIP Number:				Last Called PSTN Number:			
Last VoIP Caller:				Last PSTN Caller:	,		
Last PSTN Disconnect Reason:				PSTN Activity Timer:	30000 (ms)		
Mapped SIP Port:	Call Type:						
VoIP State:	Idle			PSTN State:	Idle		
VoIP Tone:							
VoIP Peer Name:							
VoIP Peer Number:							
VoIP Call Encoder:							
VoIP Call FAX:							
VoIP Call Duration:							
VoIP Call Packets Recv:							
VoIP Call Bytes Recv:							
VoIP Call Jitter:							
VoIP Call Packets Lost:							
VoIP Call Mapped RTP Port:							

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