LINKSYS®
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Linksys Phone Adapter Configuration

Router | Voice

Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2 — User Login | basic | advanced

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-relay
Hook Flash MIME Type:	application/hook-flash	Remove Last Reg:	no
Use Compact Header:	no	Escape Display Name:	no
RFC 2543 Call Hold:	yes	Mark All AVT Packets:	yes

SIP Timer Values (sec)

SIP T1:	.5	SIP T2:	4
SIP T4:	5	SIP Timer B:	32
SIP Timer F:	32	SIP Timer H:	32
SIP Timer D:	32	SIP Timer J:	32
INVITE Expires:	240	ReINVITE Expires:	30
Reg Min Expires:	1	Reg Max Expires:	7200
Reg Retry Intvl:	30	Reg Retry Long Intvl:	1200
Reg Retry Random Delay:		Reg Retry Long Random Delay:	
Reg Retry Intvl Cap:			

Response Status Code Handling

SIT1 RSC:		SIT2 RSC:	
SIT3 RSC:		SIT4 RSC:	
Try Backup RSC:		Retry Reg RSC:	

RTP Parameters

RTP Port Min:	16384	RTP Port Max:	16482
RTP Packet Size:	0.020	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no
Stats In BYE:	no		

SDP Payload Types

NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
INFOREQ Dynamic Payload:		G726r16 Dynamic Payload:	98
G726r24 Dynamic Payload:	97	G726r32 Dynamic Payload:	2
G726r40 Dynamic Payload:	96	G729b Dynamic Payload:	99
NSE Codec Name:	NSE	AVT Codec Name:	telephone-event
G711u Codec Name:	PCMU	G711a Codec Name:	PCMA
G726r16 Codec Name:	G726-16	G726r24 Codec Name:	G726-24
G726r32 Codec Name:	G726-32	G726r40 Codec Name:	G726-40
G729a Codec Name:	G729a	G729b Codec Name:	G729ab
G723 Codec Name:	G723		

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

Undo All Changes | Submit All Changes

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Line Enable:	yes		

Streaming Audio Server (SAS)

SAS Enable:	no	SAS DLG Refresh Intvl:	30
SAS Inbound RTP Sink:			

NAT Settings

NAT Mapping Enable:	yes	NAT Keep Alive Enable:	yes
NAT Keep Alive Msg:	$NOTIFY	NAT Keep Alive Dest:	$PROXY

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 Port:	5060	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

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Xfer When Hangup Conf:	yes	Conference Bridge URL:	
Conference Bridge Ports:	3		

Proxy and Registration

Proxy:	pbx.pbxspace.eu		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	300	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Voice Mail Server:		Mailbox Subscribe Expires:	2147483647

Subscriber Information

Display Name:	0002*21	User ID:	0002*21
Password:	password	Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			

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	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	yes
Call Back Serv:	yes	Three Way Call Serv:	yes
Three Way Conf Serv:	yes	Attn Transfer Serv:	yes
Unattn Transfer Serv:	yes	MWI Serv:	yes
VMWI 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:	G711a	Silence Supp Enable:	no
Use Pref Codec Only:	no	Silence Threshold:	medium
G729a Enable:	no	Echo Canc Enable:	yes
G723 Enable:	no	Echo Canc Adapt Enable:	yes
G726-16 Enable:	no	Echo Supp Enable:	yes
G726-24 Enable:	no	FAX CED Detect Enable:	no
G726-32 Enable:	no	FAX CNG Detect Enable:	no
G726-40 Enable:	no	FAX Passthru Codec:	G711a
DTMF Process INFO:	no	FAX Codec Symmetric:	yes
DTMF Process AVT:	yes	FAX Passthru Method:	NSE
DTMF Tx Method:	AVT	DTMF Tx Mode:	Strict
FAX Process NSE:	yes	Hook Flash Tx Method:	None
FAX Disable ECAN:	no	Release Unused Codec:	yes
FAX Enable T38:	no	FAX T38 Redundancy:	1
FAX Tone Detect Mode:	caller or callee		

Dial Plan

Dial Plan:	(x.)		
Enable IP Dialing:	no	Emergency Number:	

FXS Port Polarity Configuration

Idle Polarity:	Forward	Caller Conn Polarity:	Forward
Callee Conn Polarity:	Forward		

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