LINKSYS®
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Linksys Telephone Configuration

Info | System | SIP | Provisioning | Regional | Phone | Ext 1 | Ext 2 | Ext 3 | Ext 4 | User

User Login basic | advanced
Personal Directory Call History

System Information

DHCP:	Enabled	Current IP:	192.168.0.108
Host Name:	SipuraSPA	Domain:	socal.rr.com
Current Netmask:	255.255.255.0	Current Gateway:	192.168.0.1
Primary DNS:	192.168.0.1		
Secondary DNS:			

Product Information

Product Name:	SPA-941	Serial Number:	4KM00K302730
Software Version:	5.1.8	Hardware Version:	1.0.4(6623)
MAC Address:	000E083DFD1E	Client Certificate:	Installed
Licenses:	None		

Phone Status

Current Time:	1/1/2003 12:37:11	Elapsed Time:	00:30:35
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	1183	Broadcast Bytes Recv:	79976
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	0	RTP Bytes Sent:	0
RTP Packets Recv:	0	RTP Bytes Recv:	0
SIP Messages Sent:	1	SIP Bytes Sent:	486
SIP Messages Recv:	1	SIP Bytes Recv:	364
External IP:			

Ext 1 Status

Registration State:	Registered	Last Registration At:	1/1/2003 12:06:36
Next Registration In:	1735 s	Message Waiting:	No
Mapped SIP Port:			

Ext 2 Status

Registration State:	Not Registered	Last Registration At:	
Next Registration In:		Message Waiting:	No
Mapped SIP Port:			

Ext 3 Status

Registration State:	Not Registered	Last Registration At:	
Next Registration In:		Message Waiting:	No
Mapped SIP Port:			

Ext 4 Status

Registration State:	Not Registered	Last Registration At:	
Next Registration In:		Message Waiting:	No
Mapped SIP Port:			

Line 1 Call 1 Status

Call State:	Idle	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	

Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 1 Call 2 Status

Call State:	Idle	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 2 Call 1 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 2 Call 2 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 3 Call 1 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 3 Call 2 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 4 Call 1 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Line 4 Call 2 Status

Call State:	Disabled	Tone:	None
Encoder:		Decoder:	
Type:		Remote Hold:	
Callback:		Peer Name:	
Peer Phone:		Duration:	
Packets Sent:		Packets Recv:	
Bytes Sent:		Bytes Recv:	
Decode Latency:		Jitter:	
Round Trip Delay:		Packets Lost:	
Packet Error:		Mapped RTP Port:	

Downloaded Ring Tone

Status:	Idle
Ring Tone 1:	Not Installed
Ring Tone 2:	Not Installed

Undo All Changes   Submit All Changes

User Login   basic | advanced

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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
Remove Last Reg:	no	Use Compact Header:	no
Escape Display Name:	no	SIP-B Enable:	no
Talk Package:	no	Hold Package:	no
Conference Package:	no	Notify Conference:	no
RFC 2543 Call Hold:	yes	Random REG CID On Reboot:	no

SIP Timer Values (sec)

SIP T1:	.5	SIP T2:	4
SIP T4:	5	SIP Timer B:	16
SIP Timer F:	16	SIP Timer H:	16
SIP Timer D:	16	SIP Timer J:	16
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:		Sub Min Expires:	10
Sub Max Expires:	7200	Sub Retry Intvl:	10

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:	0.030	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no
Symmetric RTP:	no	Stats In BYE:	no

SDP Payload Types

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	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

Linksys Key System Parameters

Linksys Key System:	no	Multicast Address:	224.168.168.168:6061
Force LAN Codec:	none		

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Personal Directory Call History

General

Station Name:		Voice Mail Number:	
Text Logo:			

Line Key 1

Extension:	1	Short Name:	$USER
Share Call Appearance:	private		

Line Key 2

Extension:	Disabled	Short Name:	$USER
Share Call Appearance:	private		

Line Key 3

Extension:	Disabled	Short Name:	$USER
Share Call Appearance:	private		

Line Key 4

Extension:	Disabled	Short Name:	$USER
Share Call Appearance:	private		

Miscellaneous Line Key Settings

SCA Line ID Mapping:	Vertical First	SCA Barge-In Enable:	no

Line Key LED Pattern

Idle LED:		Remote Undefined LED:	
Local Seized LED:		Remote Seized LED:	
Local Progressing LED:		Remote Progressing LED:	
Local Ringing LED:		Remote Ringing LED:	
Local Active LED:		Remote Active LED:	
Local Held LED:		Remote Held LED:	
Register Failed LED:		Disabled LED:	
Registering LED:		Call Back Active LED:	

Supplementary Services

Conference Serv:	yes	Attn Transfer Serv:	yes
Blind Transfer Serv:	yes	DND Serv:	yes
Block ANC Serv:	yes	Call Back Serv:	yes
Block CID Serv:	yes	Secure Call Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Paging Serv:	yes
Call Park Serv:	yes	Call Pick Up Serv:	yes
ACD Login Serv:	no	Group Call Pick Up Serv:	yes

ACD Ext:	1	Service Annc Serv:	no

Ring Tone

Ring1:	n=Classic-1;w=3;c=1
Ring2:	n=Classic-2;w=3;c=2
Ring3:	n=Classic-3;w=3;c=3
Ring4:	n=Classic-4;w=3;c=4
Ring5:	n=Simple-1;w=2;c=1
Ring6:	n=Simple-2;w=2;c=2
Ring7:	n=Simple-3;w=2;c=3
Ring8:	n=Simple-4;w=2;c=4
Ring9:	n=Simple-5;w=2;c=5
Ring10:	n=Office;w=4;c=1

Audio Input Gain (dB)

Handset Input Gain:	0	Headset Input Gain:	0
Speakerphone Input Gain:	0		

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General

Line Enable:	yes		

Share Line Appearance

Share Ext:	private	Shared User ID:	
Subscription Expires:	3600		

NAT Settings

NAT Mapping Enable:	no	NAT Keep Alive Enable:	no
NAT Keep Alive Msg:	$NOTIFY	NAT Keep Alive Dest:	$PROXY

Network Settings

SIP TOS/DiffServ Value:	0x68	SIP CoS Value:	3
RTP TOS/DiffServ Value:	0xb8	RTP CoS Value:	6
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:	no
Referor Bye Delay:	4	Refer-To Target Contact:	no
Referee Bye Delay:	0	SIP Debug Option:	none
Refer Target Bye Delay:	0	Sticky 183:	no
Auth INVITE:	no		

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Message Waiting:	no	Auth Page:	no
Default Ring:	1	Auth Page Realm:	
Conference Bridge URL:		Auth Page Password:	
Mailbox ID:		Voice Mail Server:	
State Agent:		CFWD Notify Serv:	no
CFWD Notifier:			

Proxy and Registration

Proxy:	sip.skype.com	Use Outbound Proxy:	no
Outbound Proxy:	sip.skype.com	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	yes
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal

Subscriber Information

Display Name:		User ID:	99051000130763
Password:	*************	Use Auth ID:	no
Auth ID:			
Mini Certificate:			
SRTP Private Key:			

Audio Configuration

Preferred Codec:	G711u	Use Pref Codec Only:	no
G729a Enable:	yes	G723 Enable:	yes
G726-16 Enable:	yes	G726-24 Enable:	yes
G726-32 Enable:	yes	G726-40 Enable:	yes
Release Unused Codec:	yes	DTMF Process AVT:	yes
Silence Supp Enable:	no	DTMF Tx Method:	Auto

Dial Plan

Dial Plan:	(*x.|**x.|*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx| xxxxxxxxxxxx
Enable IP Dialing:	yes

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