

## **CallVoip Telefonie – configuratie Linksys WIP-310**

De Linksys WIP310 is een WiFi VoIP telefoon: een telefoon die u aanmeldt op een draadloos netwerk, waar u ook bent. Als het netwerk VoIP-verkeer toestaat en doorlaat, dan kunt u met deze telefoon via VoIP bellen en gebeld worden.

### **De Linksys WIP-310 configureren**

Het IP-adres dat de WIP-310 van de netwerkrouter toegewezen krijgt vindt u op de telefoon onder menu [info]. Toets dat IP-adres in op uw netwerkbrowser (internet explorer, Safari, etc.). De configuratiepagina van de WIP-310 verschijnt.

De Linksys WIP-310 heeft dezelfde gebruikersinterface als de overige Linksys VoIP producten. Op de volgende pagina vindt u enkele schermprints.

**General**

Line Enable:

**Share Line Appearance**

Share Ext:  Shared User ID:   
 Subscription Expires:

**NAT Settings**

NAT Mapping Enable:  NAT Keep Alive Enable:   
 NAT Keep Alive Msg:  NAT Keep Alive Dest:

**Network Settings**

SIP TOS/DiffServ Value:  SIP CoS Value:   
 RTP TOS/DiffServ Value:  RTP CoS Value:   
 Network Jitter Level:  Jitter Buffer Adjustment:

**SIP Settings**

SIP Transport:  SIP Port:   
 SIP 100REL Enable:  EXT SIP Port:   
 Auth Resync-Reboot:  SIP Proxy-Require:   
 SIP Remote-Party-ID:  Referor Bye Delay:   
 Refer-To Target Contact:  Referee Bye Delay:   
 SIP Debug Option:  Refer Target Bye Delay:   
 Sticky 183:  Set G729 annexb:

**Call Feature Settings**

Blind Attn-Xfer Enable:  MOH Server:   
 Message Waiting:  Auth Page:   
 Auth Page Realm:  Conference Bridge URL:   
 Auth Page Password:  Mailbox ID:   
 Voice Mail Server:  State Agent:   
 CFWD Notify Serv:  CFWD Notifier:

**Proxy and Registration**

Proxy:  Use Outbound Proxy:   
 Outbound Proxy:  Use OB Proxy In Dialog:   
 Register:  Make Call Without Reg:   
 Register Expires:  Ans Call Without Reg:   
 Use DNS SRV:  DNS SRV Auto Prefix:   
 Proxy Fallback Intvl:  Proxy Redundancy Method:

**Subscriber Information**

Display Name:  User ID:   
 Password:  Use Auth ID:   
 Auth ID:   
 Mini Certificate:   
 SRTP Private Key:

**Audio Configuration**

Preferred Codec:  Use Pref Codec Only:   
 G729a Enable:  G726-32 Enable:   
 Release Unused Codec:  DTMF Process AVT:   
 Silence Supp Enable:  DTMF Tx Method:

**Dial Plan**

Dial Plan:   
 Enable IP Dialing:



**Info**

System | User

[Admin Login](#)

### System Information

Connection Type:	DHCP	Current IP:	192.168.21.209
Host Name:	CiscoPhone	Domain:	
Current Netmask:	255.255.255.0	Current Gateway:	192.168.21.1
Primary DNS:	192.168.22.1	Secondary DNS:	192.168.23.1
Primary NTP Server:		Secondary NTP Server:	
TFTP Server:			

### Product Information

Product Name:	WIP310	Serial Number:	ACC132102S4
Software Version:	5.0.11(10301355)	Hardware Version:	R03
MAC Address:	001B53FF6613	Client Certificate:	Installed
Licenses:	None		

### Phone Status

Current Time:	12/31/1969 18:16:21	Elapsed Time:	00:08:30
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	0	Broadcast Bytes Recv:	0
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	210	RTP Bytes Sent:	33600
RTP Packets Recv:	206	RTP Bytes Recv:	32960
SIP Messages Sent:	41	SIP Bytes Sent:	5329
SIP Messages Recv:	14	SIP Bytes Recv:	6581
External IP:			

### Ext 1 Status

Registration State:	Registered	Last Registration At:	12/31/1969 18:07:55
Next Registration In:	83 s	Message Waiting:	No
Mapped SIP Port:			

### EXT 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:	
Media Loopback:			

### EXT 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:	
Media Loopback:			