

Configurare Linksys RTP300



Conectati adaptorul la placa de retea a calculatorului dumneavoastra folosind unul din cele 4 porturi ethernet, apoi alimentati echipamentul.

Interfata web de adminstrare este accesibila la adresa <http://192.168.15.1> folosind username'ul "admin" si parola "admin".

The image shows the web interface of the Linksys RTP300 router. The top header is blue with the Linksys logo and "A Division of Cisco Systems, Inc." on the left, and "Firmware Version: 3.1.17" on the right. Below the header is a black navigation bar with "Login" on the left, "Broadband Router with 2 Phone Ports" in the center, and "RTP300" on the right. The main content area has a black header "Username&Password" on the left. In the center, there are two input fields: "Username:" and "Password:". At the bottom, there are two buttons: "Log In" and "Cancel". The bottom right corner features the Cisco Systems logo.

Dupa ce ati efectuat setarile de retea corespunzatoare si ati conectat adaptorul la internet accesati meniul Voice, tabul Line 1 (presupunand ca aveti telefonul conectat la portul Phone 1) Verificati ca sunteti logat in modul Admin si meniul de administrare selectat este Advanced View.

Efectuati setarile conform exemplului de mai jos si apasati Save Settings

LINKSYS
A Division of Cisco Systems, Inc. Firmware Version: 3.1.17

Broadband Router with 2 Phone Ports **rtp300?**

Voice | Setup | Security | Access Restrictions | Applications & Gaming | Administration | Status | Voice

Info | System | SIP | Provisioning | Regional | **Line 1** | Line 2 | User 1 | User 2

Advanced View (switch to basic view) User Login

NAT Settings

Line Enable:

NAT Mapping Enable: NAT Keep Alive Enable:

NAT Keep Alive Msg: NAT Keep Alive Dest:

Network Settings

SIP ToS/DiffServ Value: SIP CoS Value: [0-7]

RTP ToS/DiffServ Value: RTP CoS Value: [0-7]

SIP Settings

SIP Port: SIP 100REL Enable:

EXT SIP Port: Auth Resync-Reboot:

SIP Proxy-Require: SIP Remote-Party-ID:

SIP Debug Option: RTP Log Intvl:

Restrict Source IP: Referor Bye Delay:

Refer Target Bye Delay: Referee Bye Delay:

Refer-To Target Contact: Sticky 183:

Call Feature Settings

Blind Attn-Xfer Enable: MOH Server:

Xfer When Hangup Conf: Conference Bridge URL:

Conference Bridge Ports:

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:

Voice Mail Server:	<input type="text"/>	Mailbox Subscribe Expires:	<input type="text" value="2147483647"/>
Subscriber Information			
Display Name:	<input type="text" value="account id"/>	User ID:	<input type="text" value="account id"/>
Password:	<input type="text" value="voip password"/>	Use Auth ID:	<input type="text" value="yes"/>
Auth ID:	<input type="text" value="account id"/>		
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 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"/>	VMM Serv:	<input type="text" value="yes"/>
Speed Dial Serv:	<input type="text" value="yes"/>	Secure Call Serv:	<input type="text" value="yes"/>
Referral Serv:	<input type="text" value="yes"/>	Feature Dial Serv:	<input type="text" value="yes"/>
Audio Configuration			
Preferred Codec:	<input type="text" value="G729a"/>	Silence Supp Enable:	<input type="text" value="no"/>
Use Pref Codec Only:	<input type="text" value="yes"/>	Silence Threshold:	<input type="text" value="medium"/>
G729a Enable:	<input type="text" value="yes"/>	Echo Canc Enable:	<input type="text" value="yes"/>
G723 Enable:	<input type="text" value="yes"/>	Echo Canc Adapt Enable:	<input type="text" value="yes"/>
Echo Supp Enable:	<input type="text" value="yes"/>	FAX CED Detect Enable:	<input type="text" value="yes"/>
G726-32 Enable:	<input type="text" value="yes"/>	FAX CNG Detect Enable:	<input type="text" value="yes"/>
G726-40 Enable:	<input type="text" value="yes"/>	FAX Passthru Codec:	<input type="text" value="G711u"/>
DTMF Process INFO:	<input type="text" value="yes"/>	FAX Codec Symmetric:	<input type="text" value="yes"/>
DTMF Process AVT:	<input type="text" value="yes"/>	FAX Passthru Method:	<input type="text" value="NSE"/>
DTMF Tx Method:	<input type="text" value="Auto"/>	FAX Process NSE:	<input type="text" value="yes"/>
Hook Flash Tx Method:	<input type="text" value="None"/>	FAX Disable ECAN:	<input type="text" value="no"/>
Release Unused Codec:	<input type="text" value="yes"/>	FAX Enable T38:	<input type="text" value="yes"/>
Dial Plan			
Dial Plan:	<input type="text" value="[*xx]0[2-9]xxxxxxxxx[2-3]xxxxxxxx1xx[2-9]xxxxxS0]xxxxx"/>		
Enable IP Dialing:	<input type="text" value="yes"/>	Emergency Number:	<input type="text"/>
FXS Port Polarity Configuration			
Idle Polarity:	<input type="text" value="Forward"/>	Caller Conn Polarity:	<input type="text" value="Forward"/>
Callee Conn Polarity:	<input type="text" value="Forward"/>		

Save Settings Cancel Settings



In cazul in care adaptorul dumneavoastra se afla in spatele unui NAT este necesar sa specificati stun server'ul. Aceasta optiune se afla in meniul Voice sub tabul SIP.

Hook Flash MIME Type: application/hook-fl		Type: application/sdp	
Use Compact Header: no		Remove Last Reg: no	
		Escape Display Name: no	
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
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
Stats In BYE:	no		
SDP Payload Types			
NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
INFOREQ Dynamic Payload:		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
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:	yes	STUN Test Enable:	no
STUN Server:	stun.netmaster.ro	EXT IP:	
EXT RTP Port Min:		NAT Keep Alive Intvl:	15
Save Settings		Cancel Settings	



**datele corespunzatoare campurilor account id si voip password din exemplul de mai sus se gasesc in interfata web de adminstrare a numarului dumneavoastra de telefon disponibile la www.smartcall.ro/login