

Cisco SPA112 \ Cisco SPA122

1. Setup occurs in the web interface. To log into it, enter the gateway IP address in your browser's address bar. You can find the gateway IP address by connecting a phone to it and dialing ****110#. The default username and password are both **admin**.



2. Open Voip>SIP and enter the following RTP port range:

RTP Port Min: 1024

RTP Port Max: 65455

RTP Parameters

RTP Port Min:	<input type="text" value="1024"/>	RTP Port Max:	<input type="text" value="65455"/>
RTP Packet Size:	<input type="text" value="0.030"/>	Max RTP ICMP Err:	<input type="text" value="0"/>
RTCP Tx Interval:	<input type="text" value="0"/>	No UDP Checksum:	<input type="text" value="no"/>
Stats In BYE:	<input type="text" value="yes"/>		

SDP Payload Types

NSE Dynamic Payload:	<input type="text" value="100"/>	AVT Dynamic Payload:	<input type="text" value="101"/>
INFOREQ Dynamic Payload:	<input type="text"/>	G726r32 Dynamic Payload:	<input type="text" value="2"/>
G729b Dynamic Payload:	<input type="text" value="99"/>	EncapRTP Dynamic Payload:	<input type="text" value="112"/>
RTP-Start-Loopback Dynamic Payload:	<input type="text" value="113"/>	RTP-Start-Loopback Codec:	<input type="text" value="G711u"/>
NSE Codec Name:	<input type="text" value="NSE"/>	AVT Codec Name:	<input type="text" value="telephone-event"/>
G711u Codec Name:	<input type="text" value="PCMU"/>	G711a Codec Name:	<input type="text" value="PCMA"/>
G726r32 Codec Name:	<input type="text" value="G726-32"/>	G729a Codec Name:	<input type="text" value="G729a"/>
G729b Codec Name:	<input type="text" value="G729ab"/>	EncapRTP Codec Name:	<input type="text" value="encaprtsp"/>

NAT Support Parameters

Handle VIA received:	<input type="text" value="no"/>	Handle VIA rport:	<input type="text" value="no"/>
Insert VIA received:	<input type="text" value="no"/>	Insert VIA rport:	<input type="text" value="no"/>
Substitute VIA Addr:	<input type="text" value="no"/>	Send Resp To Src Port:	<input type="text" value="no"/>
STUN Enable:	<input type="text" value="no"/>	STUN Test Enable:	<input type="text" value="no"/>
STUN Server:	<input type="text"/>	EXT IP:	<input type="text"/>
EXT RTP Port Min:	<input type="text"/>	NAT Keep Alive Intvl:	<input type="text" value="15"/>

Linksys Key System Parameters

Linksys Key System:	<input type="text" value="no"/>	Multicast Address:	<input type="text" value="224.168.168.168:6061"/>
Key System Auto Discovery:	<input type="text" value="yes"/>	Key System IP Address:	<input type="text"/>
Force LAN Codec:	<input type="text" value="none"/>		

3. Open Voip => Line1 and enter the following information:

NAT Mapping Enable: no

NAT Keep AliveEnable: yes

Proxy: *login.smarttelplus.eu*

Register Expires: 300

Display Name: your SIP number (111111, for example) from Settings => User management

UserID: your SIP number (111111, for example) from Settings => User management

Password: your password for the SIP number from Settings => User management

Preferred Codec: G711a

Second Preferred Codec: G711u

G.729a Enable: yes

To make sure that incoming calls are transferred correctly in NPBX, list the following parameters:

DTMF Tx Method: auto

DTMF Tx Mode: normal

General			
Line Enable:		yes ▾	
Streaming Audio Server (SAS)			
SAS Enable:		no ▾	
SAS Inbound RTP Sink:		<input type="text"/>	
		SAS DLG Refresh Intvl:	
		<input type="text" value="30"/>	
NAT Settings			
NAT Mapping Enable:		no ▾	
NAT Keep Alive Msg:		\$NOTIFY	
		NAT Keep Alive Enable:	
		yes ▾	
		NAT Keep Alive Dest:	
		\$PROXY	
Network Settings			
SIP ToS/DiffServ Value:		<input type="text" value="0x68"/>	
RTP ToS/DiffServ Value:		<input type="text" value="0xb8"/>	
Network Jitter Level:		high ▾	
		SIP CoS Value:	
		<input type="text" value="3"/> [0-7]	
		RTP CoS Value:	
		<input type="text" value="6"/> [0-7]	
		Jitter Buffer Adjustment:	
		yes ▾	
SIP Settings			
SIP Transport:		UDP ▾	
SIP 100REL Enable:		no ▾	
Auth Resync-Reboot:		yes ▾	
SIP Remote-Party-ID:		yes ▾	
SIP Debug Option:		none ▾	
Restrict Source IP:		no ▾	
Refer Target Bye Delay:		<input type="text" value="0"/>	
Refer-To Target Contact:		no ▾	
		SIP Port:	
		<input type="text" value="5060"/>	
		EXT SIP Port:	
		<input type="text"/>	
		SIP Proxy-Require:	
		<input type="text"/>	
		SIP GUID:	
		no ▾	
		RTP Log Intvl:	
		<input type="text" value="0"/>	
		Referor Bye Delay:	
		<input type="text" value="4"/>	
		Referee Bye Delay:	
		<input type="text" value="0"/>	
		Sticky 183:	
		no ▾	
Proxy and Registration			
Proxy:		<input type="text" value="login.smarttelplus.eu"/>	
Outbound Proxy:		<input type="text"/>	
Use Outbound Proxy:		no ▾	
Register:		yes ▾	
Register Expires:		<input type="text" value="300"/>	
Use DNS SRV:		no ▾	
Proxy Fallback Intvl:		<input type="text" value="3600"/>	
Mailbox Subscribe URL:		<input type="text"/>	
		Use OB Proxy In Dialog:	
		yes ▾	
		Make Call Without Reg:	
		no ▾	
		Ans Call Without Reg:	
		no ▾	
		DNS SRV Auto Prefix:	
		no ▾	
		Proxy Redundancy Method:	
		Normal ▾	
		Mailbox Subscribe Expires:	
		<input type="text" value="2147483647"/>	
Subscriber Information			
Display Name:		<input type="text" value="111111"/>	
Password:		<input type="text" value="*****"/>	
Auth ID:		<input type="text"/>	
		User ID:	
		<input type="text" value="111111"/>	
		Use Auth ID:	
		no ▾	
		Resident Online Number:	
		<input type="text"/>	
Supplementary Service Subscription			
Call Waiting Serv:		no ▾	
Block ANC Serv:		yes ▾	
Cfwd All Serv:		yes ▾	
Cfwd No Ans Serv:		yes ▾	
Cfwd Last Serv:		yes ▾	
Accept Last Serv:		yes ▾	
CID Serv:		yes ▾	
Call Return Serv:		yes ▾	
Call Back Serv:		yes ▾	
Three Way Conf Serv:		yes ▾	
Unattn Transfer Serv:		yes ▾	
VMWI Serv:		yes ▾	
Secure Call Serv:		yes ▾	
Feature Dial Serv:		yes ▾	
		Block CID Serv:	
		yes ▾	
		Dist Ring Serv:	
		yes ▾	
		Cfwd Busy Serv:	
		yes ▾	
		Cfwd Sel Serv:	
		yes ▾	
		Block Last Serv:	
		yes ▾	
		DND Serv:	
		yes ▾	
		CWCID Serv:	
		yes ▾	
		Call Redial Serv:	
		yes ▾	
		Three Way Call Serv:	
		yes ▾	
		Attn Transfer Serv:	
		yes ▾	
		MWI Serv:	
		yes ▾	
		Speed Dial Serv:	
		yes ▾	
		Referral Serv:	
		yes ▾	
		Service Announcement Serv:	
		no ▾	
Audio Configuration			
Preferred Codec:		G711a ▾	
Third Preferred Codec:		Unspecified ▾	
G729a Enable:		yes ▾	
G726-32 Enable:		no ▾	
FAX V21 Detect Enable:		yes ▾	
FAX CNG Detect Enable:		yes ▾	
FAX Codec Symmetric:		yes ▾	
FAX Passthru Method:		NSE ▾	
FAX Process NSE:		yes ▾	
FAX Disable ECAI:		no ▾	
DTMF Tx Strict Hold Off Time:		<input type="text" value="70"/>	
Hook Flash Tx Method:		None ▾	
FAX T38 ECM Enable:		no ▾	
Symmetric RTP:		no ▾	
		Second Preferred Codec:	
		G711u ▾	
		Use Pref Codec Only:	
		no ▾	
		Silence Supp Enable:	
		no ▾	
		Silence Threshold:	
		medium ▾	
		Echo Canc Enable:	
		yes ▾	
		FAX Passthru Codec:	
		G711a ▾	
		DTMF Process INFO:	
		yes ▾	
		DTMF Process AVT:	
		yes ▾	
		DTMF Tx Method:	
		Auto ▾	
		DTMF Tx Mode:	
		Normal ▾	
		FAX Enable T38:	
		no ▾	
		FAX T38 Redundancy:	
		1 ▾	
		FAX Tone Detect Mode:	
		caller or callee ▾	

To save your changes, click **Submit**.