Configure Network Address Translation (NAT) Support Parameters on SPA300/SPA500 Series IP Phones

Objective

Session Initiation Protocol (SIP) is a signaling protocol used to create, manage, and terminate sessions in an IP based network. SIP is a mechanism for call management. It also allows for the establishment of user location, provides for feature negotiation so that all of the participants in a session can agree on the features to be supported among them, and allows for changes to be made to features of a session while it is in progress.

Network Address Translation (NAT) modifies the IP address while it crosses through a traffic routing device in IP packet headers. It provides security to hide the internal IP address from view.

The objective of this document is to explain how to configure NAT Support Parameters on the SPA300 and SPA500 Series IP Phones.

Applicable Devices

- SPA300 Series IP Phone

- SPA500 Series IP Phone

NAT Support Parameters Configuration

Note: On the actual SPA300 or SPA500 Series IP Phone, to set signaling protocol as **SIP**, use the navigation keys to go to **Device Administration > Call Control Settings > Signaling Protocol SIP.**

Step 1. Log in to the web configuration utility and choose **Admin Login > Advanced > Voice > SIP**. The *SIP Parameters* page opens:

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:	User
SIP Accept Language:	English	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
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	Mark All AVT Packets:	yes
SIP TCP Port Min:	5060	SIP TCP Port Max:	5080
CTI Enable:	no	Caller ID Header:	PAID-RPID-FROM
SRTP Method:	x-sipura	Hold Target Before REFER:	no
Dialog SDP Enable:	no	Keep Referee When REFER Failed:	no
Display Diversion Info:	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

Step 2. Scroll down to the *NAT Support Parameters* area.

SDP Payload Types			
AVT Dynamic Payload:	101	INFOREQ Dynamic Payload:	
G726r32 Dynamic Payload:	2	G729b Dynamic Payload:	99
EncapRTP Dynamic Payload:	112	RTP-Start-Loopback Dynamic Payload:	113
RTP-Start-Loopback Codec:	G711u	AVT Codec Name:	telephone-event
G711u Codec Name:	PCMU	G711a Codec Name:	PCMA
G726r32 Codec Name:	G726-32	G729a Codec Name:	G729a
G729b Codec Name:	G729ab	G722 Codec Name:	G722
EncapRTP Codec Name:	encaprtp		

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:	10.1.1.11	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
Key System Auto Discovery:	no	Key System IP Address:	
Force LAN Codec:	none		

Step 3. Choose **Yes** or **No** from the *Handle VIA Received* drop-down list. If you choose Yes the IP Phone will use the IP address when it gets any received parameters in a VIA header. The default is No.

Step 4. Choose **Yes** or **No** from the *Handle VIA rport* drop-down list. If you choose Yes the IP Phone will use the UDP port when it gets any rport parameters in a VIA header. The default is No.

Step 5. Choose **Yes** or **No** from the *Insert VIA received* drop-down list. If you choose Yes it will insert the received parameters in the VIA header when there is a difference between received-from IP and VIA sent-by IP. The default is No.

Step 6. Choose **Yes** or **No** from the *Insert VIA rport* drop-down list. If you choose Yes it will insert the rport parameters in the VIA header when there is a difference between received-from IP and VIA sent-by IP. The default is No.

Step 7. Choose **Yes** or **No** from the *Substitute VIA Addr* drop-down list. If you choose Yes a NAT-mapped IP will be used in the VIA header. The default is No.

Step 8. Choose **Yes** or **No** from the *Send Resp To Src Port* drop-down list. If you choose Yes the responses will be sent to the request source port instead of the VIA sent-by port. The default is No.

Step 9. Choose **Yes** or **No** from the *STUN Enable* drop-down list. If you choose Yes, STUN will be used to discover NAT mapping. The default is No.

Step 10. Choose **Yes** or **No** from the *STUN Test Enable* drop-down list. If you choose Yes the IP Phone will operate as a NAT-type operation. The IP Phone will contact with the STUN server and report a warning header in all Register requests. The default is No.

Step 11. Enter the IP address or the domain name of the STUN server in the *STUN Server* field. This helps NAT to map through the connection with the STUN server.

Step 12. Enter the external IP address in the *EXT IP* field to use in the place of the actual IP address of the IP Phone. The default is blank.

Step 13. Enter the minimal external port map number in the *EXT RTP Port Min* field to use in the place of the private UDP port of the IP Phone. The default is blank.

Step 14. Enter the maximum interval in seconds between two packets used to keep alive in the *NAT Keep Alive Intvl* field. The default is 15.

Step 15. Click **Submit All Changes** to save the settings.