Configure Real-time Transport Protocol (RTP) 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.

Real-time Transport Protocol (RTP) is an internet protocol to carry data which has real-time properties. It is a standard format to transmit real-time data such as audio, video.

The objective of this document is to explain the configuration of Real-time Transport Protocol (RTP) Parameters on SPA300 and SPA500 Series IP Phones.

Applicable Devices

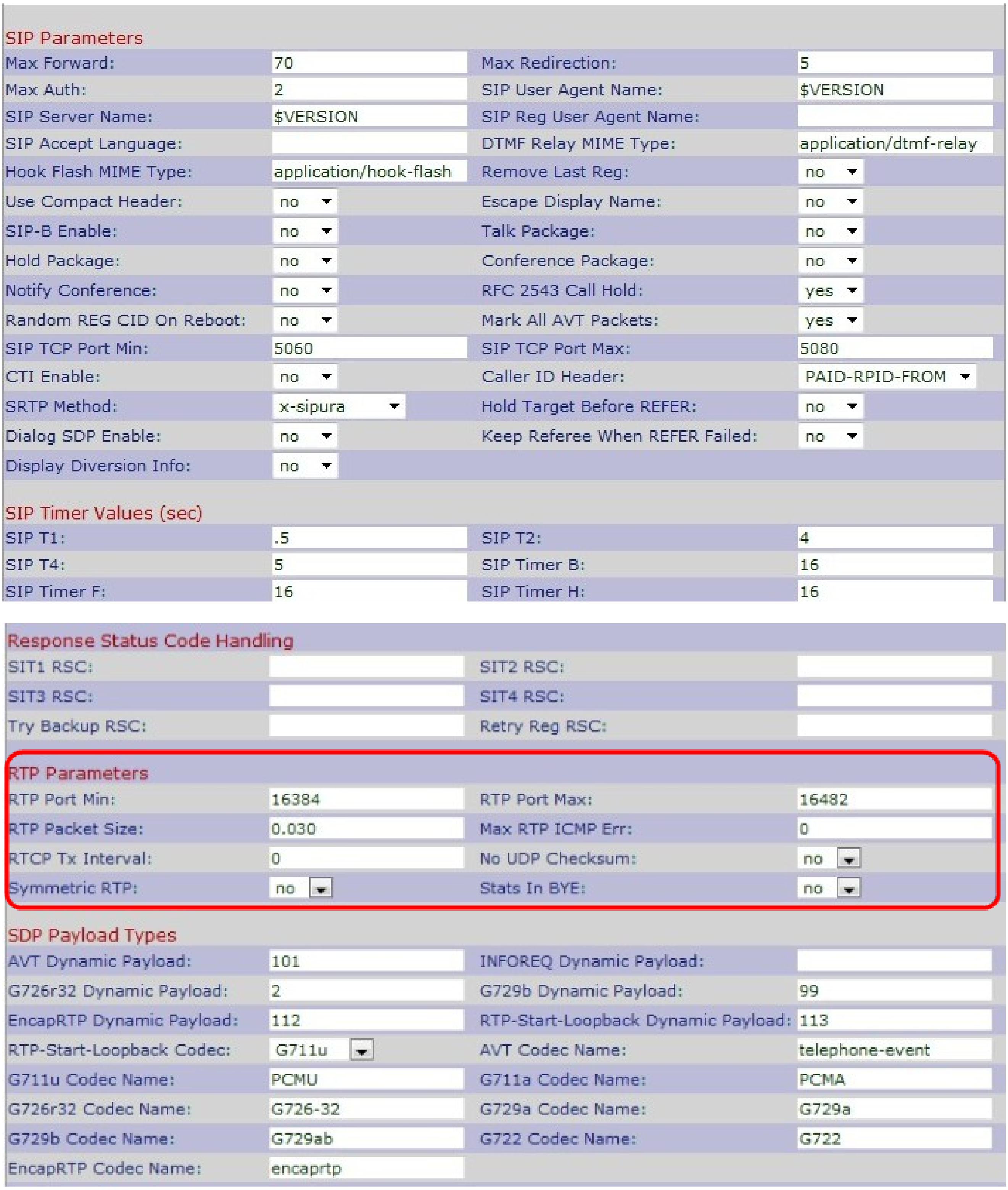
• SPA300 Series IP Phone

• SPA500 Series IP Phone

RTP Parameters Configuration

Note: On the actual SPA300 or SPA500 Series IP Phone 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:	
SIP Accept Language:		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

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:	
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		

Step 2. Scroll down to the RTP Parameters area.

Step 3. Enter the minimum port number in the *RTP Port Min* field. It is the minimum range which contains at least ten even number ports for transmission and reception. The default is 16384.

Step 4. Enter the maximum port number in the *RTP Port Max* field. It is the maximum range which contains at least ten even number ports for transmission and reception. The default is 16482.

Step 5. Enter the size of RTP packet in the *RTP Packet Size* field. The range is from 0.01 to 0.16. The default is 0.030.

Step 6. Enter the number of successive Internet Control Message Protocol (ICMP) errors allowed before the termination of the IP Phone in the *Max RTP ICMP Err* field. ICMP is a internet protocol which is used to send network error message. The default is 0.

Step 7. Enter the interval to send out sender reports of the Real-Time Transport Control Protocol (RTCP) on an active connection in the *RTCP Tx Interval* field. The range is from 0 to 255 seconds. The defaults is 0.

Step 8. Choose **Yes** or **No** from the *No UDP Checksum* drop-down list. If you choose **Yes,** the IP Phone will calculate the UDP header checksum for SIP messages.

Step 9. Choose **Yes** or **No** from the *Symmetric RTP* drop-down list. If you choose **Yes,** the RTP packets will be sent to the source address and if you choose **No** the RTP packets will be sent to the destination address. The default is **No**.

Step 10. Choose **Yes** or **No** from the *Stats in BYE* drop-down list. If you choose **Yes** the P-RTP-Stat header will be sent in response to a BYE message. The default is **No**.

Step 11. Click **Submit All Changes** to save the settings.