

# Configure Session Description Protocol (SDP) Payload Types 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.

Session Description Protocol (SDP) is a standard representation to describe media streams for multi-media sessions. SDP itself does not carry media but negotiates between the end points of media type and format. The SDP Payloads are used for outbound calls only. If there are inbound calls the IP Phone follows the caller payload type.

The objective of this document is to show you the configuration of SDP Payload Types on SPA300 and SPA500 Series IP Phones.

## Applicable Devices

- SPA300 Series IP Phone
- SPA500 Series IP Phone

## SDP Payload Types 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

Step 2. Scroll down to the SDP Payload Types area.

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:	encaprtpt		

Step 3. Enter the number on which the sender and receiver must agree for the session event in the *AVT Dynamic Payload* field. The range is from 96 to 127. The default is 101.

Step 4. Enter the codec number which is used to send an SIP message in the *INFOREQ Dynamic Payload* field. The best range is from 96 to 27. The default is blank.

**Note:** The INFOREQ Dynamic Payload number should match with the network or other party number which is configured to enable the Dynamic Payload.

Step 5. Enter the RTP Payload Type Number in the *G726r16 Dynamic Payload* field. This is

the number which represents the G.726r16 codec transmitted packet. The range is from 96 to 127. The default is 98.

**Note:** The G26r16 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 6. Enter the RTP Payload Type Number in the *G726r24 Dynamic Payload* field. This is the number which represents the G.726r24 codec transmitted packet. The range is from 96 to 127. The default is 97.

**Note:** The G26r24 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 7. Enter the RTP Payload Type Number in the *G726r32 Dynamic Payload* field. This is the number which represents the G.726r32 codec transmitted packet. The range is from 0 to 268435455. The default is 2.

Step 8. Enter the RTP Payload Type Number in the *G726r40 Dynamic Payload* field. This is the number which represents the G.726r40 codec transmitted packet. The range is from 0 to 268435455. The default is 2.

Step 9. Enter the RTP Payload Type Number in the *G729b Dynamic Payload* field. This is the number which represents the G.729b codec transmitted packet. The range is from 0 to 268435455. The default is 99.

Step 10. Enter the Encapsulated RTP Dynamic Payload Type in the *EncapRTP Dynamic Payload field*. The range is from 0 to 268435455. The default is 112.

Step 11. Enter the value which indicates the RTP-Start-Loopback in the *RTP-Start-Loopback Dynamic* field. In RTP-Start-Loopback, a network element blocks the media of the loopback-source until the loopback-mirror starts the transmission of the packet. The default is 113.

Step 12. Choose the appropriate RTP-Start-Loopback Codec to convert an analog voice signal to digital encoded voice signal from the *RTP-Start-Loopback Codec* drop-down list. The default is G711u.

- G711u — This is a Pulse Code Modulation (PCM) scheme. This uses mu-law codec which improves signal-to-noise-ratio without the requirement of more additional data. It is used in the United States and Japan.
- G711a — This is a Pulse Code Modulation (PCM) scheme. This uses A-law codec, and is used in most of the countries of the world.
- G726-32 — This is an Adaptive Different Pulse Code Modulation (ADPCM) scheme. This uses both a-law and mu-law, and helps to reduce the bandwidth.
- G729a — This is an extension of G729 which uses Algebraic Code Excited Linear Prediction (ACELP) to simplify the complexity and cut the high cost of G729. It requires less computational power than G729.
- G722 — This is a Sub-Band Adaptive Different Pulse Code Modulation (SB-ADPCM) scheme, which provides 7kHz wide band audio data rates to improve the speech quality.

Step 13. Enter the name of the Audio Video Transport (AVT) codec in the *AVT Codec Name* field. The default is telephone-event.

Step 14. Enter the name of the G711u codec in the *G711u Codec Name* field. This is a Pulse Code Modulation (PCM) scheme which uses mu-law codec to improve signal-to-noise-

ratio without the requirement of more additional data. It is used in the United States and Japan. The default is Pulse Code Modulation mu-law (PCMU).

Step 15. Enter the name of the G711a codec in the *G711a Codec Name* field. This is a Pulse Code Modulation (PCM) scheme which uses A-law codec. It is used in most of the countries of the world. The default is Pulse Code Modulation A-law (PCMA).

Step 16. Enter the name of the G726r16 codec in the *G726r16 Codec Name* field. This is an Adaptive Differential Pulse Code Modulation (ADPCM) scheme which uses 16 kbit per second. The default is G726-16.

**Note:** The G726r16 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 17. Enter the name of the G726r24 codec in the *G726r24 Codec Name* field. This is an Adaptive Differential Pulse Code Modulation (ADPCM) scheme which uses 24 kbit per second. The default is G726-24.

**Note:** G726-24 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 18. Enter the name of the G726r32 code in the *G726r32 Codec Name* field. This is an Adaptive Differential Pulse Code Modulation (ADPCM) scheme which uses 32 kbit per second. The default is G726-32.

Step 19. Enter the name of the G726r40 codec in the *G726r40 Codec Name* field. This is an Adaptive Differential Pulse Code Modulation (ADPCM) scheme which uses 40 kbit per second. The default is G726-40.

**Note:** G726-40 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 20. Enter the name of the G729a codec in the *G729a Codec Name* field. This is an extension of G729 which uses Algebraic Code Excited Linear Prediction (ACELP) to simplify the complexity and cut the high cost of G729. It requires less computational power than G729. The default is G729a.

Step 21. Enter the name of the G729b codec in the *G729b Codec Name* field. This is an extension of G729 to provide support to wideband speech and audio. The default is G729ab.

Step 22. Enter the name of the G722 codec in the *G722 Codec Name* field. This is a codec that gives support of 7 kHz wideband audio. The default is G722.

Step 23. Enter the name of the G723 code in the *G723 Codec Name* field. This is an extension of G721 which uses Adaptive Differential Pulse Code Modulation (ADPCM) to provide 300 Hz to 3400 Hz voice quality. The default is G723.

**Note:** G723 Dynamic Payload is not applicable to the SPA525G and SPA525G2.

Step 24. Enter the name of the EncapRTP codec in the *EncapRTP Codec Name* field. This is the encapsulated Real-Time Protocol name. The default is encaprtpt.

Step 25. Click **Submit All Changes** to save the settings.