

Configure Session Initiation Protocol (SIP) 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.

The objective of this document is to show you the configuration of SIP Parameter on SPA300 and SPA500 Series IP Phone.

Applicable Devices

- SPA300 Series IP Phone
- SPA500 Series IP Phone

Linksys Key Configuration

Note: On the actual SPA300 or SPA500 Series IP Phone set signaling protocol as **SIP**, use 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* 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. Enter the number of proxies or gateways that can forward the request to the next downstream server in the *Max Forward* field. This is the value which indicates the remaining number of times the request message is allowed to be forwarded. The range is from 0 to 255. The initial value is 70.

Step 3. Enter the number of times an invite can be redirected to avoid an infinite loop in the *Max Redirection* field. The default is 5.

Step 4. Enter the maximum number of times a request might be challenged in the *Max Auth* field. The range is from 0 to 255. The default is 2.

Step 5. Enter the User-Agent header used in outbound requests in the *SIP User Agent Name* field. The default is \$VERSION. If empty, the header is not included.

Step 6. Enter the server header used in responses to inbound responses in the *SIP Server Name* field. The default is \$VERSION.

Step 7. Enter the User-Agent name used in a register request in the *SIP Reg User Agent Name* field. If not specified, the SIP User Agent Name is used for the register request.

Step 8. Enter the name of the preferred language in the *SIP Accept Language* field for reason phrases, session descriptions, or status responses which are carried as message bodies in the response. If blank, the header is not included and the server assumes that all languages are acceptable to the client. The default is blank.

Step 9. Enter the DTMF relay MIME in the *DTMF Relay MIME Type* field. MIME Type is used in a SIP INFO message to signal a DTMF event. This parameter must match with the service provider. The default is application/dtmf-relay.

Step 10. Enter the Hook Flash MIME in the *Hook Flash MIME Type* field. MIME Type used in a SIPINFO message to signal a hook flash event.

Step 11. Choose **Yes** or **No** from the *Remove Last Reg* drop-down list. If you choose Yes, it will remove the last registration used before registering a new one if the value is different. The default is No.

Step 12. Choose **Yes** or **No** from the *Use Compact Header* drop-down list. If you choose Yes, the Cisco IP phone uses compact SIP headers in outbound SIP messages. If you choose no, the Cisco SPA IP phones use normal SIP headers. The default is No.

Step 13. Choose **Yes** or **No** from the *Escape Display Name* drop-down list. If you choose Yes, it will enclose the configured Display Name string in a pair of double quotes for outbound SIP messages.

Step 14. Choose **Yes** or **No** from the *SIP-B Enable* drop-down list. If you choose Yes, it will enable SIP for Business (supports Sylanro call flows) call features.

Step 15. Choose **Yes** or **No** from the *Talk Package* drop-down list. If you choose Yes, it will enable support for the BroadSoft Talk Package that lets users answer or resume a call by clicking a button in an external application. The default is No.

Step 16. Choose **Yes** or **No** from the *Hold Package* drop-down list. If you choose Yes, it will enable support for the BroadSoft Hold Package, which lets users place a call on hold by clicking a button in an external application. The default is No.

Step 17. Choose **Yes** or **No** from the *Conference Package* drop-down list. If you choose Yes, it will enable support for the BroadSoft Conference Package that enables users to start a conference call by clicking a button in an external application. The default is No.

Step 18. Choose **Yes** or **No** from the *Notify Conference* drop-down list. If you choose Yes, Cisco SPA IP phones send out a NOTIFY with event conference when starting a conference call (with the BroadSoft Conference Package). The default is No.

Step 19. Choose **Yes** or **No** from the *RFC 2543 Call Hold* drop-down list. If you choose Yes, Cisco SPA IP phones include Session Description Protocol (SDP) syntax c=0.0.0.0 when sending a SIP re-INVITE to a peer to hold the call. The default is Yes.

Step 20. Choose **Yes** or **No** from the *Random REG CID On Reboot* drop-down list. If you choose Yes, Cisco SPA IP phones use a different random call-ID for registration after the next software reboot. The default is No.

Step 21. Choose **Yes** or **No** from the *Mark ALL AVT Packets* drop-down list. If you choose Yes, all audio video transport (AVT) tone packets (encoded for redundancy) have the marker bit set. The default is Yes.

Step 22. Enter the lowest TCP port number that can be used for SIP sessions in the *SIP TCP Port Min* field. The default is 5060.

Step 23. Enter the highest TCP port number that can be used for SIP sessions in the *SIP TCP Port Max* field. The default is 5080.

Step 24. Choose **Yes** or **No** from the *Keep Referee When REFER Failed* drop-down list. If you choose Yes, the phone immediately handles NOTIFY sipfrag messages.

Step 25. Choose **Yes** or **No** from the *CTI Enable* drop-down list. If you choose Yes, it will enable the computer telephony integration (CTI), where a computer can act as a call center handling all sorts of incoming and outgoing communications, including phone calls,

faxes, and text messages. The CTI interface allows a third-party application to control and monitor the state of a Cisco IP phone and, for example, initiate or answer a call by clicking a mouse on a PC. The default is no.

Note: CTI must be enabled on the Cisco SPA300 Series or Cisco SPA500 Series IP phones for an attached Cisco Attendant Console to properly monitor the IP phone line status.

Step 26. Choose the specific caller ID header from the *Caller ID Header* drop-down list. The default is PAID-RPID-FROM.

Step 27. Choose the specific SRTP method from the *SRTP Method* drop-down list. SRTP is the method to use for secure real time transport protocol. The default is x-sipura.

Step 28. Choose **Yes** or **No** from the *Hold Target Before REFER* drop-down list. If you choose Yes, it will control whether to hold the call leg with the transfer target before sending REFER to the transferee when initiating a fully attended call transfer (where the transfer target has answered). The default is No.

Step 29. Choose **Yes** or **No** from the *Dialog SDP Enable* drop-down list. If you choose Yes, you will get simplified Notify message xml.

Step 30. Choose **Yes** or **No** from the *Display Diversion Info* drop-down list. If you choose Yes, it displays the Diversion Header information, if it exists, in the INVITE message.

Step 31. Click **Submit All Changes** to save the settings.