Session Initiation Protocol (SIP) Parameters Configuration on SPA8000

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. SIP allows the users to send packets that consist of more than one stream. Applications and programs like video conferencing, instant message, and media streaming have packets that consist of more than one stream. This document explains how to configure SIP parameters on a SPA8000.

Applicable Device

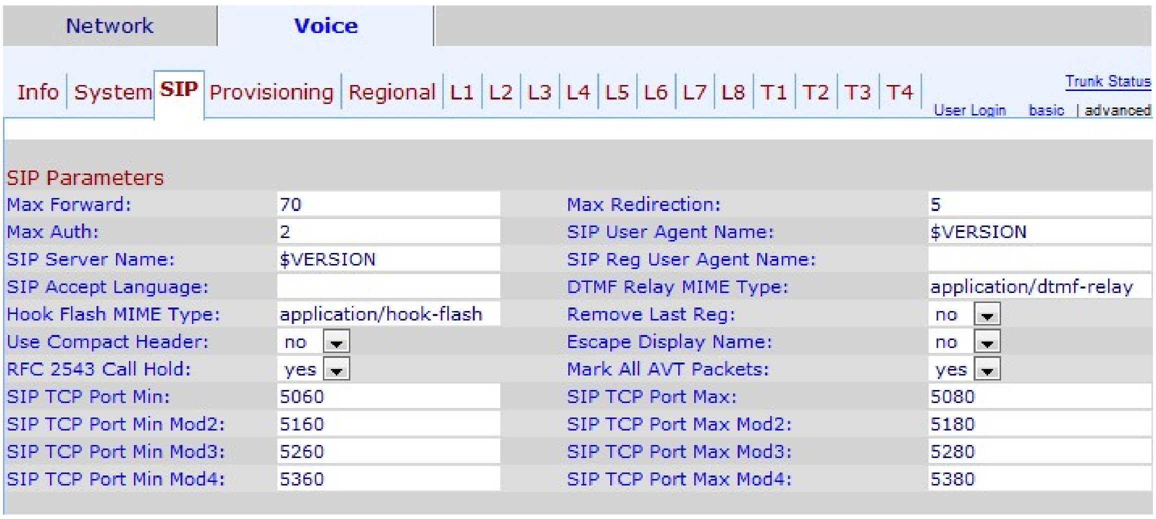
- SPA8000

Software Version

- 6.1.12

SIP Parameters Configuration

Step 1. Log in to the web configuration utility as an administrator and choose **Advanced > Voice > SIP**. The *SIP* page opens:



Step 2. Enter the SIP maximum forward value in the Max Forward field. This is the limit on the number of proxies or gateways that can forward the request to the next downstream server. The default Max Forward value is 70.

Step 3. Enter the maximum number of times that an invite can be redirected to avoid an infinite loop in a network in the Max Redirection field. The default is 5. An invite is when a user is invited to take part in a call.

Step 4. Enter the maximum number of times that a request may be challenged in the Max Auth field. The Max Auth value can range from 0 to 255.

Step 5. Enter the User-Agent header used in outbound requests in the SIP User Agent Name field. The default User Agent Name is $VERSION. If this field is left empty, then the header will not include any value.

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

Step 7. Enter the User-Agent name to be used in a register request in the SIP Reg User Agent Name field.

Step 8. Enter the preferred language header to be used in the SIP Accept Language field. This field is used in requests to indicate the preferred languages for reason phrases, or status responses carried as message bodies in the response. If no Accept Language header field is present, the server should assume all languages are acceptable to the client.

Step 9. Enter the MIME type used in a SIP info message to signal a DTMF event in the DTMF Relay MIME Type field. The default is application/dtmf-relay. Multi-Purpose Internet Mail Extensions (MIME) is an Internet standard that increases the capabilities of email. Dual Tone Multi Frequency (DTMF) is the signal that a phone generates when a soft key is pressed to establish connection.

Step 10. Enter the MIME type used in a SIP info message to signal a hook flash event in the Hook Flash MIME Type field. The default is application/hook-flash. Hook flash simulates hanging up the phone and then quickly picking it up again.

Step 11. Choose **yes** to remove the last registration used before registering a new one if the value is different from the Remove Last Reg drop-down list. The default is yes.

Step 12. Choose **yes** to use compact SIP headers in outbound SIP messages instead of normal SIP headers from the Use Compact Header drop-down list. The default is yes. The use of compact headers minimize the size of the message.

Step 13. Choose **no** to keep your display name private from the Escape Display Name drop-down list. The default is no. This feature allow the administrator to hide the name (ID) of the user of SIP.

Step 14. Choose **yes** to configure the type of call hold from the RFC 2543 Call Hold drop-down list. The default is no.

Step 15. Choose **yes** from the Mark All AVT Packets drop-down list to have all AVT tone packets have a marker bit set. If you choose no, then only the first packet has a marker bit set for each DTMF event. The marker bit is used to identify the AVT packet.

Step 16. Enter the lowest TCP port number that can be used for SIP sessions in the SIP TCP Port Min field.

Step 17. Enter the highest TCP port number that can be used for SIP sessions in the SIP TCP Port Max field.

Note: There are up to 3 extra SIP TCP Port Min/Max Mod number fields that can be used for redundancy.

Step 18. (Optional) Enter the lowest TCP port number that can be used for SIP sessions in the SIP TCP Port Min Mod(2/3/4) field.

Step 19. (Optional) Enter the lowest TCP port number that can be used for SIP sessions in the SIP TCP Port Max Mod(2/3/4) field.

Step 20. Click **Submit All Changes** to save the configuration.