

Configure Response Status Code Handling 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.

This article explains the configuration of response status code handling on SPA300 and SPA500 Series IP Phones.

Applicable Devices

- SPA300 Series IP Phone
- SPA500 Series IP Phone

Response Status Code 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* 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 ▾ |
| S RTP 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: | <input type="text"/> | SIT2 RSC: | <input type="text"/> |
| SIT3 RSC: | <input type="text"/> | SIT4 RSC: | <input type="text"/> |
| Try Backup RSC: | <input type="text"/> | Retry Reg RSC: | <input type="text"/> |

| RTP Parameters | | | |
|-------------------|------------------------------------|-------------------|------------------------------------|
| RTP Port Min: | <input type="text" value="16384"/> | RTP Port Max: | <input type="text" value="16482"/> |
| RTP Packet Size: | <input type="text" value="0.030"/> | Max RTP ICMP Err: | <input type="text" value="0"/> |
| RTCP Tx Interval: | <input type="text" value="0"/> | No UDP Checksum: | <input type="text" value="no"/> |
| Symmetric RTP: | <input type="text" value="no"/> | Stats In BYE: | <input type="text" value="no"/> |

| SDP Payload Types | | | |
|---------------------------|--|-------------------------------------|--|
| AVT Dynamic Payload: | <input type="text" value="101"/> | INFOREQ Dynamic Payload: | <input type="text"/> |
| G726r32 Dynamic Payload: | <input type="text" value="2"/> | G729b Dynamic Payload: | <input type="text" value="99"/> |
| EncapRTP Dynamic Payload: | <input type="text" value="112"/> | RTP-Start-Loopback Dynamic Payload: | <input type="text" value="113"/> |
| RTP-Start-Loopback Codec: | <input type="text" value="G711u"/> | AVT Codec Name: | <input type="text" value="telephone-event"/> |
| G711u Codec Name: | <input type="text" value="PCMU"/> | G711a Codec Name: | <input type="text" value="PCMA"/> |
| G726r32 Codec Name: | <input type="text" value="G726-32"/> | G729a Codec Name: | <input type="text" value="G729a"/> |
| G729b Codec Name: | <input type="text" value="G729ab"/> | G722 Codec Name: | <input type="text" value="G722"/> |
| EncapRTP Codec Name: | <input type="text" value="encaprtsp"/> | | |

Step 2. Scroll down to the Response Status Code Handling area.

Step 3. Enter a SIP response status code for the appropriate Special Information Tone (SIT) in the SIT1 RSC field. It is an alternative to the recorder tone which is played when an error occurs as a caller makes an unbound call. The default is blank.

Step 4. Enter a SIP response status code that will result in the SIT2 Tone being played in the SIT2 RSC field. It is an alternative to the recorder tone which is played when an error occurs as a caller makes an unbound call. The default is blank.

Step 5. Enter a SIP response status code that will result in the SIT3 Tone being played in the SIT3 RSC field. It is an alternative to the recorder tone which is played when an error occurs as a caller makes an unbound call. The default is blank.

Step 6. Enter a SIP response status code that will result in the SIT4 Tone being played in the SIT4 RSC field. It is an alternative to the recorder tone which is played when an error occurs as a caller makes an unbound call. The default is blank.

Step 7. Enter a SIP response code that retries a backup server for the current request in the Try Backup RSC field. The default is blank.

Step 8. Enter the interval to wait (in seconds) before the device retries registration after the failure for the duration of the last registration in the Retry Reg RSC field. The default is blank.

Step 9. Click **Submit All Changes** to save the settings.