# Configuring SIP Settings

To establish system-wide SIP settings, use tools located in the **General-SIP** directory.

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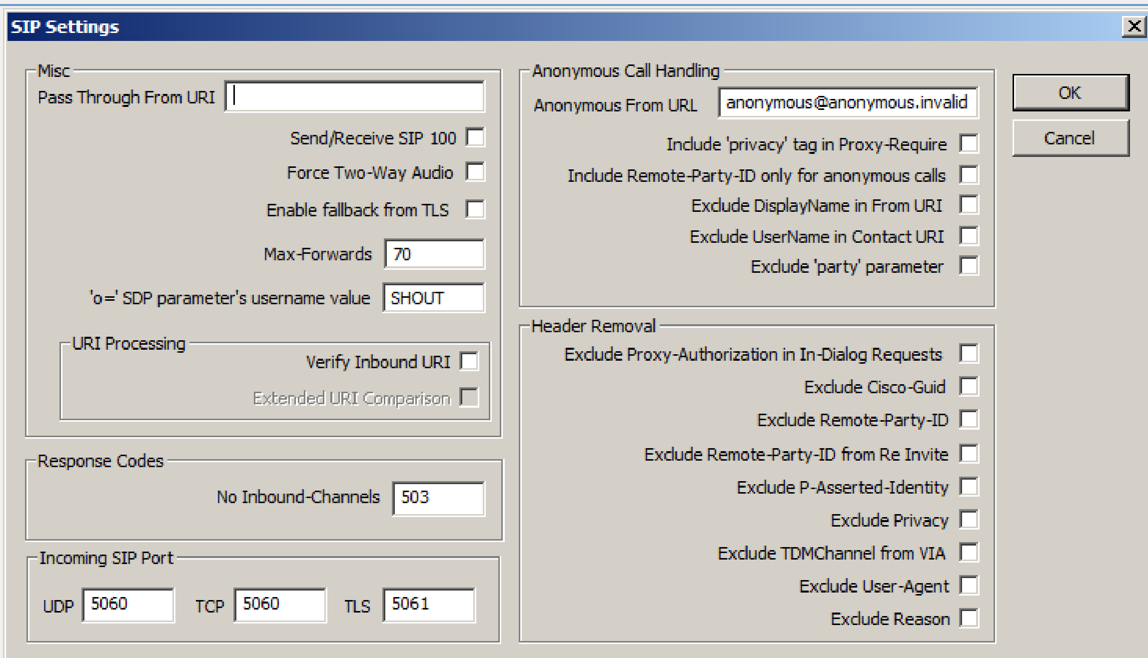
| **View Tables** | Select the **SIP** subdirectory for the VX node and view contents at the **SIP Settings** screen. |
| **Modify Settings** | Double-click on an entry in the **SIP Setting** screen to present the **SIP Settings** dialog box. |

**Edit SIP Settings Dialog**




| Field | Description |
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| **<u>Misc</u>** | |
| **Pass Through From URI** | Specifies which From header is passed through.<br>Only if there is an exact match (case insensitive) this header will be carried from the incoming to outgoing except for the Tag parameter.<br>From header is a special case since VX reads, understands and interprets the From header. If the From header needs to be carried from the incoming INVITE to the outgoing INVITE.<br><br>The pass through headers are applicable to Proxy-Like mode only. |
| **Send/Rcv SIP 100** | Enabling the option makes the VX send a 100 TRYING SIP message after getting an INVITE. This stops the retransmission of the INVITE. The disadvantage is that if the 180 RINGING SIP message is lost after the trying, no ringback will occur.<br>Disabling this option suppresses the sending of the 100 TRYING SIP message. This means VX continues to receive INVITES until the 180 RINGING occurs. and send INVITES until it gets the 180 RINGING. It will ignore the trying. With this ringback is guaranteed since the INVITES will be transmitted until the 180 RINGING is received. |
| **Force Two-Way Audio** | Forces two way audio for ISDN and SIP before the call connects. |
| **Enable Fallback from TLS** | Enables VX to fallback to other configured transport mechanisms on TLS validation failure. |

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| **Max-Forwards** | A non-zero value enables Max-Forwards in all the outgoing Requests. The outgoing (Invite or all other) request insert a value based on this configuration. The default value of this configuration is set to 70 based on the RFC3261's recommendation. The range of the value is 1-255. |
| **'o=' SDP parameter's username value** | The configured value is used in Username part of 'o=' SDP parameter. Default value of this field is 'SHOUT'. |
| **<u>URI Processing</u>** | |
| **Verify Inbound URI** | When this field is enabled, the Request URI is matched with Contact URI VX advertised in INVITE (outbound call leg) OR 200 OK (inbound call leg). If the comparison fails for an INVITE, message is dropped after sending out a 404 response if Trunk Groups-> SIP -> SIP Common -> Reject non Subscribers is configured to "yes". Else, the request will not be dropped. If comparison fails for any other non-ACK request, message is dropped after sending a 481 error response. This setting is disabled by default. |
| **Extended URI Comparison** | This setting is enabled if **Verify Inbound URI** is checked. When this option is enabled, every URI param which exists in both URIs is compared. User, ttl, method and maddr parameters are mandatory in both URIs, and must match for the comparison to succeed. All other URI parameters are compared *only* if they exist in both URIs. Parameters that are present in only one URI are ignored. This setting is disabled by default. |
| **<u>Response Codes</u>** | |
| **No Inbound-Channels** | This field indicates which SIP response should be sent for the Incoming SIP calls, if there are no free channels available. The valid range for this field is 400 to 699. If no value is configured, VX sends a 503 response by default. |
| **<u>Incoming SIP Port</u>** | |
| **UDP** | When this field is enabled, the configured port is used for SIP UDP connections. The valid range for this field is 1025-49999. The default value is 5060. |
| **TCP** | When this field is enabled, the configured port is used for SIP TCP connections. The valid range for this field is 1025-65535. The default value is 5060. |
| **TLS** | When this field is enabled, the configured port is used for SIP TLS connections. The valid range for this field is 1025-65535. The default value is 5061. |
| **<u>Anonymous Call Handling</u>** | |
| **Anonymous From URL** | When this field is enabled, the configured URL is sent in From header for anonymous calls. Max length of Anonymous From URL can be 255 characters. The default value of this field is 'anonymous@anonymous.invalid'. |
| **Include 'privacy' tag in Proxy-Require** | When this field is enabled, 'privacy' is included in Proxy-Require Header for Anonymous calls. |
| **Include Remote-Party-ID only for anonymous calls** | When this field is enabled, Remote-Party-ID is included only for Anonymous calls. |
| **Exclude DisplayName in From URI** | When this field is enabled, display name is excluded from From URI for Anonymous calls. |
| **Exclude UserName in Contact URI** | When this field is enabled, User Name is excluded from Contact URI for Anonymous calls. |
| **Exclude 'party' parameter** | When this field is enabled, 'party' parameter is excluded from Remote-Party-ID Header for Anonymous calls. |

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| **Header Removal** | The **Exclude** options below provide a means to exclude the specified header from the outgoing SIP INVITE requests. By default, these headers are added in the INVITE request. |
| **Exclude Proxy-Authentication in In-Dialog Requests** | When this field is enabled the Proxy-Authentication header is removed from outgoing SIP messages. |
| **Exclude Cisco-Guid** | When this field is enabled the Cisco-Guid header is removed from outgoing SIP messages. |
| **Exclude Remote-Party-ID** | When this field is enabled the Remote-Party-ID header from removed in outgoing SIP messages. |
| **Exclude Remote-Party-ID from Re Invite** | When this field is enabled the Remote-Party-ID header is removed from Re INVITE messages. |
| **Exclude P-Asserted-Identity** | When this field is enabled the P-Asserted-Identity header is removed from outgoing SIP messages. |
| **Exclude Privacy** | When this field is enabled the Privacy header is removed from outgoing SIP messages. |
| **Exclude TDMChannel from VIA** | When this field is enabled the TDM channel is removed from outgoing SIP messages. |
| **Exclude User-Agent** | When this field is enabled, User-Agent Header is removed from outgoing SIP messages. |
| **Exclude Reason** | When this field is enabled the Reason header is removed from outgoing SIP messages. |