

## Enabling/Disabling TLS/SRTP on Altigen SIP Trunks

This guide describes how to enable (or disable) TLS/SRTP on Altigen SIP Trunks. The process includes making configuration changes in MaxAdministrator and Enterprise Manager, then enabling (or disabling) TLS for your service in the Order Portal.

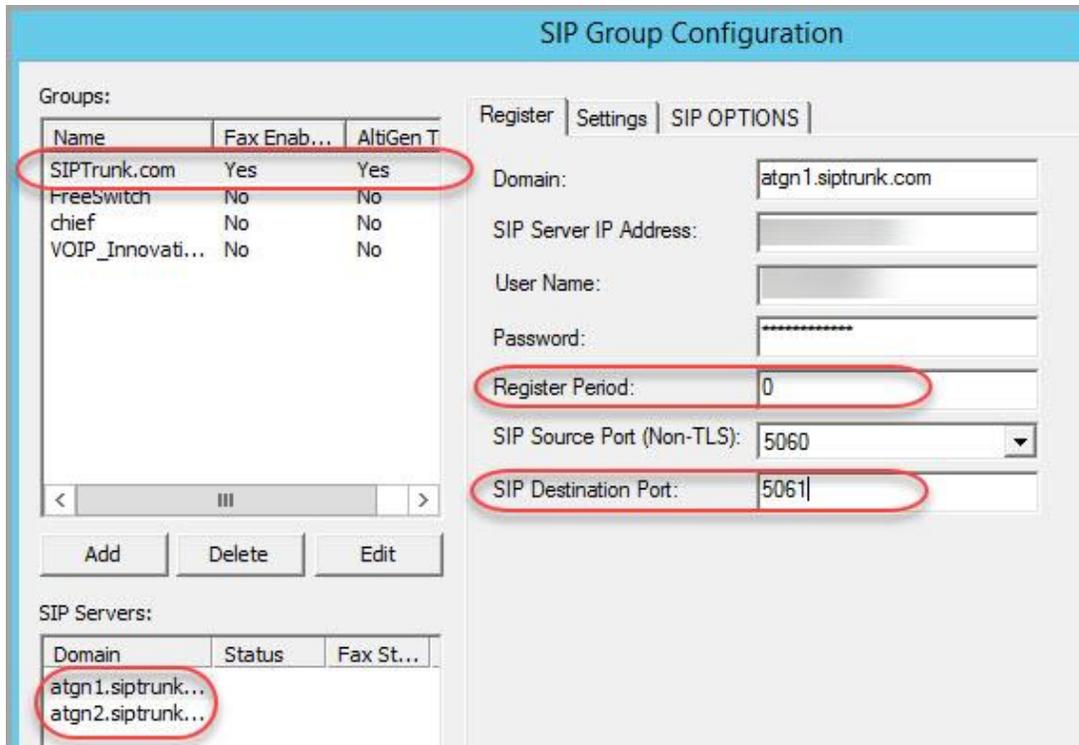


**These procedures should be performed during non-business hours, to avoid problems with active calls while the changes to the services are being made.**

You must be running MaxCS 8.5 Update 1.

### Enabling TLS on Altigen SIP Trunks

1. Ensure that TCP port 5061 is open on your firewall or router to the Altigen SIP Server addresses.
2. In MaxAdministrator, open Boards view and double-click **SIPSP**. Click **Board Configuration > SIP Group Configuration**.
3. Select the SIPTrunk.com group. Change the *Register Period* to **0** and the *SIP Destination Port* to **5061** on both SIP servers (atgn1.siptrunk.com and atgn2.siptrunk.com).



**SIP Group Configuration**

Groups:

| Name             | Fax Enab... | Altigen T |
|------------------|-------------|-----------|
| SIPTrunk.com     | Yes         | Yes       |
| FreeSwitch       | No          | No        |
| chief            | No          | No        |
| VOIP_Innovati... | No          | No        |

Register | Settings | SIP OPTIONS

Domain: atgn1.siptrunk.com

SIP Server IP Address:

User Name:

Password: \*\*\*\*\*

Register Period: 0

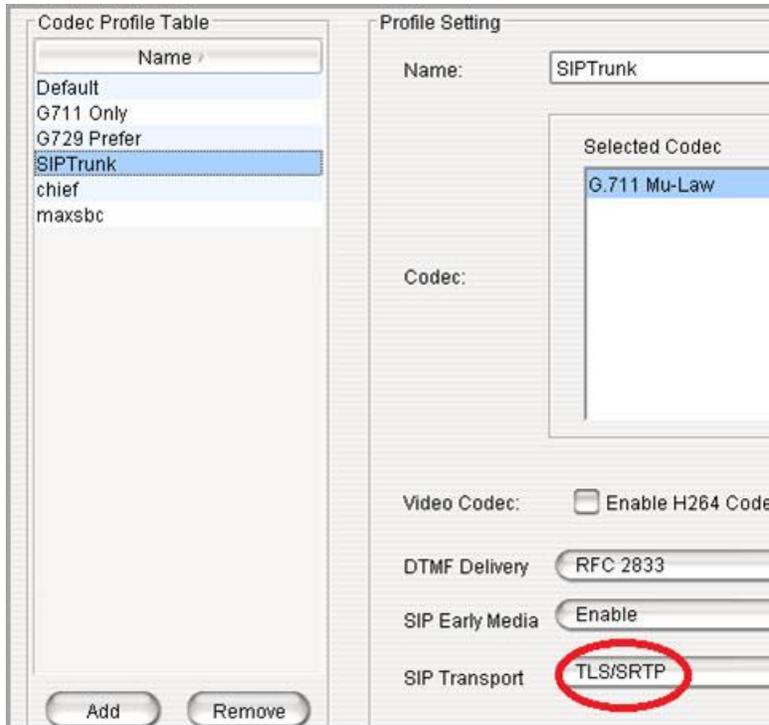
SIP Source Port (Non-TLS): 5060

SIP Destination Port: 5061

SIP Servers:

| Domain            | Status | Fax St... |
|-------------------|--------|-----------|
| atgn1.siptrunk... |        |           |
| atgn2.siptrunk... |        |           |

4. Log into Enterprise Manager and click the **Codec** button in the toolbar.
5. For the *SIPTrunk* profile, change the *SIP Transport* setting to **TLS/SRTP**.



The screenshot shows the 'Codec Profile Table' on the left and the 'Profile Setting' on the right. The 'SIPTrunk' profile is selected in the table. In the 'Profile Setting' panel, the 'SIP Transport' dropdown menu is set to 'TLS/SRTP', which is circled in red. Other settings include 'Name: SIPTrunk', 'Selected Codec: G.711 Mu-Law', 'Video Codec: Enable H264 Codec', 'DTMF Delivery: RFC 2833', and 'SIP Early Media: Enable'.

6. Repeat step 3, only this time set the *Register Period* for both *atgn1.siptrunk.com* and *atgn2.siptrunk.com* to **60**. Do not change the *SIP Destination* port setting.
7. Log into the Altigen Order portal and open the *General* tab for your service. In the *Voice SIP Trunks* section, select the checkbox *Enable TLS/SRTP on SIP Trunk*. Confirm the change when prompted.

**SIP Services**

[Download the Migration, SIP Trunk Configuration and Preliminary Testing Instructions](#)

**Voice SIP Trunk Group**

SIP Trunk Authentication : Pass through MaxCS (SIP Registration)

SIP Server IP Address : 65.254.44.194

SIP Trunk User Name :

SIP Trunk Password : \*\*\*\*\* [Hide/Show Password](#)

SIP Trunk Domain : atgn1.siptrunk.com

Second SIP Server IP Address : 74.81.71.18

Second SIP Trunk User Name :

Second SIP Trunk Password : \*\*\*\*\* [Hide/Show Password](#)

Second SIP Trunk Domain : atgn2.siptrunk.com

Enable TLS/SRTP on SIP Trunk :  [Download the TLS Configuration Guide](#)

Security PIN : [Show](#) / [Set PIN](#)

To verify your configuration:

1. Make a PSTN call through the SIP Trunk and make sure that the voice quality and connection are good.
2. On the MaxCS server, open a command line and type `spdump.exe`
3. Open the file sipman.txt. Search for the last SIP message with 183 Session Progress and make sure that the From and To header's domains are atgn1.siptrunk.com or atgn2.siptrunk.com.
4. Check if there is a RTP/SAVP tag in this SIP message. If SAVP is found in this SIP message, then TLS/SRTP is set up correctly for your SIP trunks.
5. Following is an example of the contents of the sipman.txt file.

```
2018-06-05 15:58:01,683(0) [0x0618] SipCallMan (0) recv TLS data len =
1056,org 65.254.44.194:(5061) dest (5061) after recv 0.001 seconds

SIP/2.0 183 Session Progress
Via: SIP/2.0/TLS 96.127.167.160:5061;re-
ceived=96.127.167.160;rport=50817;branch=z9hG4bK1606667_1525479919
Record-Route:
<sip:65.254.44.194;r2=on;lr=on;ftag=3501544_1525479919;vsf=AAAAAAAAAAAA
AAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAA;dlgcor=bc7.dfb1;proxy_me-
dia=yes>,<sip:65.254.44.194:5061;transport=tls;r2=on;lr=on;ftag=3501544
_1525479919;vsf=AAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAA
AAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAA;dlgcor=bc7
.dfb1;proxy_media=yes>
CSeq: 3 INVITE
To: <sip:14088873390@ >;tag=sansay1873736534rdb157035
From: <sip:4085979028@atgn1.siptrunk.com>;tag=3501544_1525479919
Call-ID: 3617960_1525479919@10.200.5.139
Contact: <sip:72.15.219.140;did=bc7.1cd2de47>
Content-Type: application/sdp
```

```
Content-Length: 314

v=0
o=Sansay-VSXi 188 1 IN IP4 65.254.44.194
s=Session Controller
c=IN IP4 65.254.44.194
t=0 0
m=audio 46022 RTP/SAVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000ff
a=fmtp:101 0-15
a=sendrecv
a=rtcp:46023
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:y5IJCvXb8xZ8oLJT8rPB-
joUxDcf+wgPGT1XjEJlB
```

## Disabling TLS on Altigen SIP Trunks

These procedures should be performed during non-business hours, to avoid problems with active calls while the changes are being made.

Follow these steps if you currently have TLS enabled and need to disable it:

1. Log into the Altigen Order portal and open the *General* tab for your service. In the *Voice SIP Trunks* section, clear the checkbox *Enable TLS/SRTP on SIP Trunks*. Confirm the change when prompted.
2. In MaxAdministrator, open Boards view and double-click **SIPSP**. Click **Board Configuration > SIP Group Configuration**. Select the *SIPTrunk.com* group. Change the *Register Period* to **0** and set the *SIP Destination Port* to **5060** on both SIP servers (atgn1.siptrunk.com and atgn2.siptrunk.com).
3. Log into Enterprise Manager and click the **Codec** button in the toolbar. For the *SIPTrunk* profile, change the *SIP Transport* setting to **UDP**.
4. Return to MaxAdministrator. Open Boards view and double-click **SIPSP**. Click **Board Configuration > SIP Group Configuration**. Select the *SIPTrunk.com* group. Change the *Register Period* back to **60**.