LT-3100/-S Configuration of SIP Trunk

How to configure the Asterisk PBX to interface to the LT-3100 Satellite Communications System.

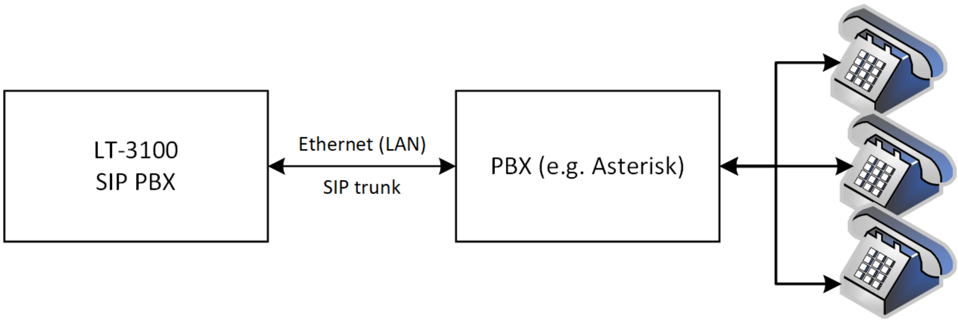

Figure 1: LT-3100/-S system (illustration of SIP trunk)

Enable a SIP account in the LT-3100/-S by using the web interface. At the menu: *configuration -> SIP* enable a SIP account. In the example below account 1100 is used (with password: 123456).

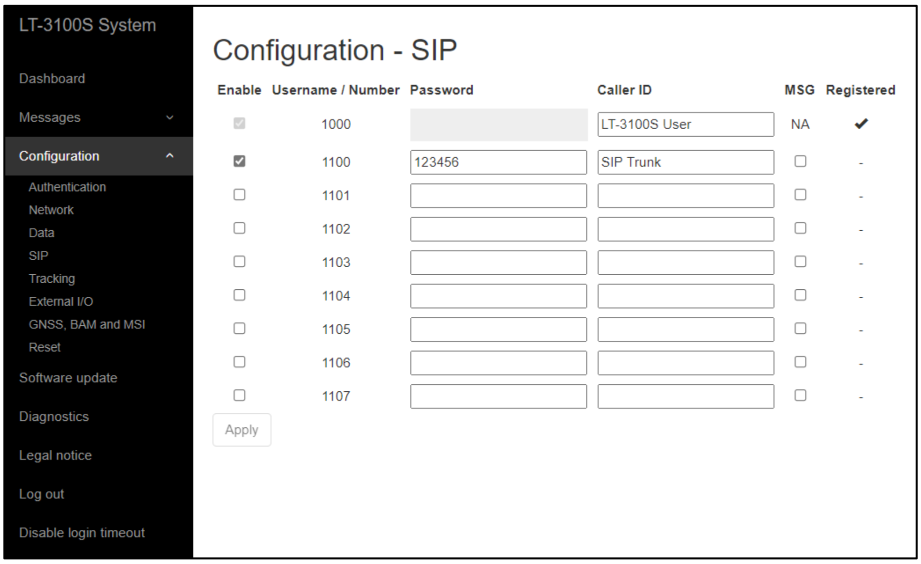

Figure 2: LT-3100/-S Web Server (Configuration -> SIP)

At the computer hosting the Asterisk application, open /etc/asterisk/sip.conf and add following:

```
; Register at LT-3100, account 1100, default IP address
register => 1100:123456@169.254.1.1

[iridium]
type=friend
username=1100
secret=123456
```

```
host=169.254.1.1
context=iridium
```

Open /etc/asterisk/extensions.conf and add following:

```
; Use this context for routing calls to Iridium
[iridium-dialout]
exten => s,1,Dial(SIP/iridium/${EXTEN},60)


; Incoming call via Iridium
[iridium]
exten => s,1,Log(NOTICE, Incoming call from ${CALLERID(all)} via Iridium)
```

References:

1 https://en.wikipedia.org/wiki/SIP_trunking
2 https://www.voip-info.org/asterisk-config-sipconf/
3 https://www.voip-info.org/asterisk-config-extensionsconf/