

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).

LT-3100S System	Con	figuration -	SIP			
Dashboard	Enable	Username / Number	Password	Caller ID	MSG	Registered
Messages v		1000		LT-3100S User	NA	
Configuration ^		1100	123456	SIP Trunk		-
Authentication		1101] 0	-
Data		1102				-
SIP		1103] 0	-
External I/O		1104				-
GNSS, BAM and MSI		1105				-
Reset		1106] _	-
Software update		1107] [] _	-
Diagnostics	Apply					
Legal notice						
Log out						
Disable login timeout						

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 <u>https://en.wikipedia.org/wiki/SIP_trunking</u>

2 https://www.voip-info.org/asterisk-config-sipconf/

3 https://www.voip-info.org/asterisk-config-extensionsconf/