VOIP Services and Performance

IP Telephony Voice over IP (VoIP) is universally replacing legacy telephony services. The VoIP capabilities of Astrolab enables us to provide a full telephony service to customers. The Figure below provides a schematic description of the elements included in the solution.

VOIP Topology

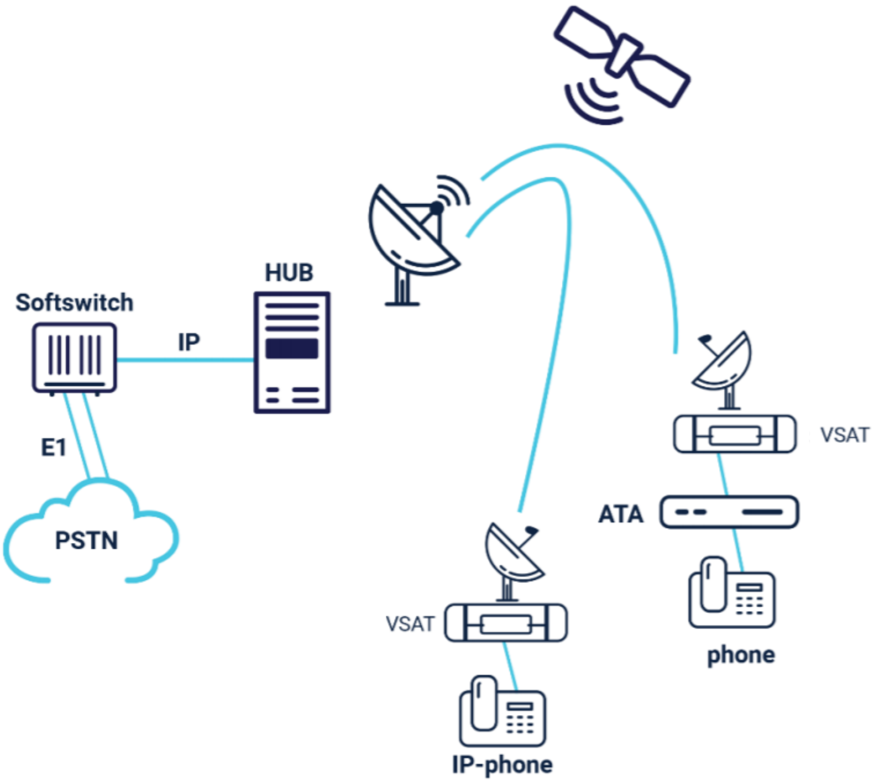

Figure 1

The main characteristics of the VoIP solution are:

- Voice quality – In order to achieve high-quality voice calls, a dedicated virtual circuit is created per call. The hub manages this circuit in such a way that it eliminates jitter and thus maintains the quality of the session.
- Codecs supported - The system supports G.729 (8Kbps), G.723.1 (5.3Kbps and 6.3Kbps), and AMR family of codecs.
- Protocols supported (for compatibility with external ATAs or IP phones):

- SIP – SIP voice calls are identified as they start. The network implementation includes call admission control (CAC). It requests BW allocation from the system when a SIP INVITE is detected. If there are enough system resources, then they are guaranteed for the whole duration of the call and a virtual circuit between the remote and the hub is created. At the end of the session the resources are returned to the system. If there are not enough system resources to start the call, the call is dropped.
- H.323, MEGACO and MGCP supported through RTP detection – the allocation of the resources and creation of the virtual connection is done when the RTP packets start to flow. The system identifies the RTP frames and allocates bandwidth based on the information inside them.
- Fax over IP – T.38 protocol for fax over IP is supported in SIP. Fax over IP over H.323 is supported using manual configuration of the FXS port.

- Strict Priority - Since VoIP runs simultaneously with other applications, the VoIP traffic will get absolute priority over any other traffic running over the VSAT.

- Efficient space segment use – In order to best use the space segment, the system employs several techniques:

  - Inbound - compresses the RTP frame header by 95% from 40 Bytes to 2 Bytes.
  - Outbound – 40% Compression of packet header (from 67 to 40 Bytes)

Any VoIP equipment that supports SIP protocol may be used. However, the implementation of different vendors may include variations. Astrolab should be consulted prior to network implementation.