VoCeM™

Voice compression and enhanced multiplexing

VoCeM provides very high compression rates for backhauled voice calls. The resulting efficiency savings enable operators to increase the number of calls over the link by 500% and higher compared to standard backhauling

VoCeM information:

- VoCeM advanced compression functions provide extremely low OPEX
- The four key stages are:
  1. Voice packets of VoIP, GSM or UMTS are transcoded using low-rate codecs
  2. Several voice packets are multiplexed in a single IP/UDP container
  3. IP-UDP-RTP headers are compressed to the absolute minimum and routed via satellite to a decompressor
  4. Voice packets are transcoded back to VoIP, GSM or UMTS and sent to the telephone networks
- Backhaul link modem management is available, with dynamic link control for most voice traffic

Advanced Compression

Vocem uses the most advanced header compression and voice codecs to provide very low bit rates, and therefore enables highly efficient use of available bandwidth

Automatic Call Detection and Compression

Vocem system integration is easy: Vocem detects the call setup and automatically starts both compression and decompression, as well as controlling the required backhaul bandwidth

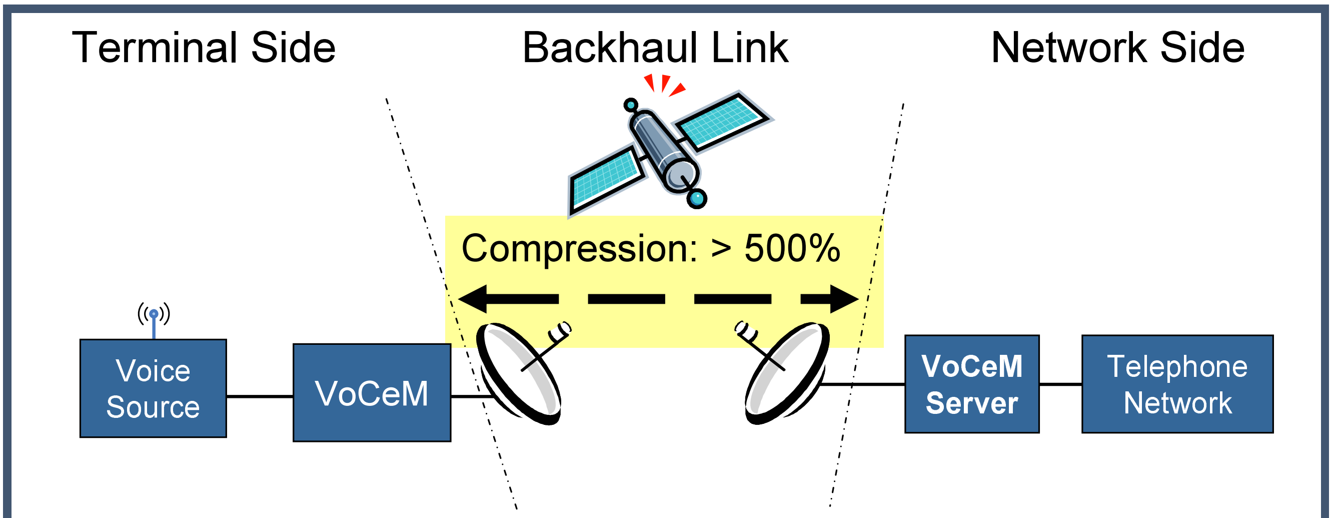
VoCeM key technical features:

- VoIP, GSM and UMTS standards are supported
- Each call uses only 6 kbit/s
- Up to 120 simultaneous calls per compressor
- VoCeM management means bandwidth is only used during data transmission
- Support for various satellite systems, including:
  - Inmarsat BGAN, Swift-64, Fleet
  - DVB-RCS
  - VSAT (Ku-band)

Key uses of VoCeM:

- Maritime: passenger and crew communications on private and commercial ships and boats
- Emergency communications for disaster recovery
- Aeronautical: VoIP or GSM communication on business and commercial aircraft
- Establishment of voice networks for remote sites, for example oil drilling stations, mines, logging camps and remote villages

VoCeM System Architecture



VoCeM solution can be integrated over most of today's satellite networks.

VoCeM runs between a remote client and a VoCeM ground server. The compression detects a new call, it then transcodes and compresses the call and forwards it to the ground server for decompression.

Each VoCeM component "learns" headers of the original packets from the other server.

Typical compression gains:

- uncompressed VoIP G.711: 96 kbit/s
- uncompressed GSM full rate: 30 kbit/s

- compressed VoIP (ILBC): 9 kbit/s (factor 10 compression gain)
- compressed GSM (AMR 4.75): 6 kbit/s (factor 5 compression gain)

- Optional transcoding into ILBC or AMR
- A wide range codecs are supported, including G.711, G.729, ILBC, GSM-FR, GSM-EFR, AMR
- VoCeM uses standard backhaul compression, for example RFC 2507 and ROHC
- VoCeM runs over either routers or NAT
- Runs on open source Linux kernels 2.6 or higher

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