

VOIP AND SATELLITE SYSTEMS

System Integrator's Perspective

By Tokuo Oishi

Satellite communication technology has been evolving for the last 15 years in many areas. There is a continuous push to provide larger and more powerful GEO satellites that can support more transponders in C, Ka and Ku bands. Modern modems using new modulation schemes such as 16 Quadrature Amplitude Modulation (QAM) have been used to improve bandwidth efficiency. Additionally, the Turbo coding is also a recent addition to improve bandwidth efficiency. The usage of VSAT systems by corporate network users has greatly increased. Broadband Internet service with VSAT offered by various system providers is a worldwide phenomenon.

Proliferation of IP technology in the corporate world and expanded usage of the Internet by the general public has led corporations to embrace VoIP technology because of the advantages of cost savings and consolidation of voice and data networks. The satellite industry is not an exception. VoIP over satellite will continue to grow as demands for communications in previously underserved areas grow and as corporations use VoIP for their satellite links.

It has been known that voice over satellite has its unique challenges and issues such as delay and associated echo. New challenges and considerations for VoIP due to the IP transmission environment and satellite links will be addressed as well as the associated solutions that the system integrators should be aware of when selecting a VoIP product vendor for their system. It is emphasized that solutions to these issues are available, but not all VoIP product vendors have the knowledge and experience with VoIP and satellite systems to provide these solutions. Proper selection of a VoIP product vendor who has the experience and expertise in satellite communications makes a huge difference in voice quality, bandwidth efficiency and system stability.

Jitter

Variations in voice packet inter-arrival time are known as jitter, generated in satellite system and IP networks due to their multiple access schemes and variations in router queue loading in time. Jitter must be absorbed at the listener side to play out the voice sample in right timing, otherwise it degrades the voice quality and if it is large, a voice conversation is not possible. For this reason dynamic jitter buffering at the listener side is required. The proper selection of buffer size for specific IP network and satellite system characteristics is important to reduce overall delay and at the same time reducing the jitter noise to a minimum.

Latency

There are many contributing factors to latency. The satellite round trip delay of about a half-second is the largest contributor to the overall delay. In addition, there are voice coding/decoding delays, IP network delays and buffering delays. Many studies have shown that satellite delay does not degrade voice quality as long as the echo is reduced to a negligible level. Using the standard VoIP signaling protocols such as H.323 and SIP, the call setup time may become considerably longer, due to the multiple interactions with a centralized call router over the satellite link. It is possible to use a faster proprietary call setup procedure provided by a few VoIP product vendors.



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Packet Loss

Packet loss may occur in IP networks due to network congestion causing routers to discard incoming packets. Voice packet loss in satellite links occur as bits in a packet may be corrupted in the path due to rain fade, for example. Voice packet loss in satellite links can be countered by proper satellite link budget and use of FEC. Proper IP network design and implementation of priority scheme for voice packets can reduce this voice packet loss. Even with proper network design, packet loss still may occur and cause disturbing noise resulting in the degradation of voice quality. Taking into consideration the human ear's characteristics, it is possible to reduce the effect of packet loss by repeating the last voice sample or inserting noise to fill the gap created by the lost packet at the receiving side.

Echo Cancellation

Echo in a voice circuit is generated at the 2-wire/4-wire hybrid (analog) at the listener side and it only affects the speaker since he is the one to hear the echo. This will occur regardless of whether the speaker is using a digital or analog device. Echo is very disturbing to the human ear therefore, most of the telephony system operators install digital echo cancellers to remove the echo. Since typical echo cancellers are designed for terrestrial delays of several tenths of a millisecond, they are not suitable for satellite links. The system integrator should check that the echo cancellation function can cancel an echo of at least half-second delay and it complies with ITU-T G.165 and G.168.

Prioritization and QoS

If data and voice packets have the same priority, the voice packet's timing is most likely to be disturbed due to the various sizes of data packets and their bursty traffic generation pattern. Even though VoIP uses a timestamp feature of RTP and buffering at the receiving side, if the timing disturbance is too large, the buffer can not absorb the timing disturbance and it results in disruptive noise as mentioned in the jitter section. It is a common practice to assign a higher priority to voice packets over data packets to reduce jitter and should be performed for satellite and IP networks. The VoIP device should have a priority queuing scheme to handle standard QoS mechanisms such as DiffServ (RFC2474)'s TOS bits.

Data Packet Fragmentation

Although voice packets are given higher priority over data packets, a large data packet may disrupt the timing of voice packets and create jitter. For example, a 1500 byte IP data packet needs 47ms transmission time using a 256 kbps satellite link. The resulting jitter must be absorbed at the receiving side. To reduce this large buffering burden, fragmentation of a large data packet to smaller sized packets may be performed.

Bandwidth Requirements per Call

The common voice compression algorithms for VoIP applications are ITU-T G. 711 PCM (64 kbps) with MOS of 4.1 as toll quality, G.729A CS-ACELP (8 kbps) with MOS of 3.7 and G.723.1 MP-MLQ (6.3 kbps) with MOS of 3.9 and G.723.1 ACELP (5.3 kbps) with MOS of 3.7. MOS of 3.6 and higher is considered acceptable for toll quality. G.711 has high voice quality and is typically used in a LAN environment. G.729A has good compression gain and robustness against packet loss and is typically used in a WAN environment and G.723.1 has high compression gain and is also used in WAN environment. It is noted that actual bandwidth consumption is larger than the pure voice-encoding rate since there is IP/UDP/RTP stack overhead and layer 2 (Ethernet/Frame Relay/ATM) overhead.

Voice Activity Detection and Comfort Noise Injection

In addition to compression of a voice call, voice activity detection (VAD) is implemented in ITU-T G.729AB (8 kbps) to further increase bandwidth efficiency. During a silent period no packet is

sent from the talker to the listener, but comfort noise is inserted to the listener side during the silent period to maintain natural voice quality. Using a typical 40% voice activity factor, a call with 8 kbps coding may be reduced to 3.2 kbps bandwidth per call resulting in a reduction of bandwidth by 60%. VAD performance varies with each VoIP product vendor; therefore the system integrator should check the VAD performance and associated voice quality carefully.

Header Compression

The compressed voice data is transported in an IP network using the IP/UDP/RTP protocol stack. The overhead is about 40 bytes (20 bytes of IP header, 8 bytes of UDP header and 12 byte of RTP header) and typical voice sample payload is 20 bytes per voice packet, resulting in more than 50% of overhead in bandwidth. The IETF addressed this issue with header compression techniques specified in RFC 2508, also called compressed RTP (cRTP), reducing 40 bytes header overhead to 2 or 4 bytes per packet depending on whether UDP checksum is sent or not. RFC 3095, Robust Header Compression (ROHC) is a method of header compression under packet loss conditions. The satellite integrator should choose a proper compression method depending on the customer network requirements.

Overall Benefit of Bandwidth Efficiency

The total length of a voice packet for typical G.729A (8kbps) is 60bytes. This packet is generated every 20ms or 50 packets per second, resulting 24 kbps of bandwidth usage. There are also proprietary voice compression methods for certain applications to save bandwidth. For example, using the proprietary VoIP system from NSGDatacom, only 10 kbps bandwidth per call is required using the proprietary 8 kbps encoding. This encoding scheme has an equivalent MOS value of G.729. It can pack 8-10 simultaneous calls per 64 kbps bandwidth, while many well known VoIP gateways can only achieve as few as 4 calls per 64 kbps bandwidth.

Customization

There is always a need for customization to fit each satellite system integrator's unique requirements. The VoIP product vendor should have overall know-how and experience to satisfy this need. The VoIP product vendor that has all the capabilities previously mentioned or knowledge in-house has the most flexibility and can provide the best customized solution.

Environmental Considerations for Hardware

VoIP over satellite solutions are often installed in harsh conditions where large temperature variations, voltage fluctuations, and dusty environments are typical. The system integrator should select the VoIP product vendor that not only claims rugged hardware, but can also provide actual field proven products, if such performance is required.

Network Management

After network installation is completed and the operational phase is started, network maintenance cost is the single most important factor in the system's life cycle. Centralized network management tool, self-diagnostic capability, redundant power supply and rapid replacement procedure with spare products are some of the necessary requirements to reduce system maintenance costs while keeping system availability high. A centralized network management tool and self-diagnostic capability should be provided by the VoIP product vendor free of charge. The system integrator should have the cost versus benefit analysis for additional items such as redundant power supply and number of spare parts at the network design phase.

Conclusions

VoIP over satellite is poised to continue growing as corporations, network providers, broadband service providers, and general telephony providers continue to implement solutions to take advantage of the cost savings and consolidation of voice and data networks. There are literally hundreds of VoIP product vendors available. However, not many VoIP product vendors know the satellite implications nor do they have the experience to provide the best fit solution to satisfy the customer's requirements. Therefore the system integrator needs to select wisely a VoIP product vendor with satellite knowledge and experience such as NSGDatacom (www.nsgdata.com).