Voice over IP (VoIP) over Satellite

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iDIRECT

With developments in IP networking in general and the Internet in particular, the ability to transmit voice calls which have been traditionally handled by the public switched telephone network (PSTN) over a lower cost IP infrastructure has become a reality. More and more users are researching VoIP capability when purchasing a connection to the Internet or simply expanding their private intranet. VoIP based telephony has also provided a very cost-effective solution for companies that have locations in many disparate regions and require connectivity back to their headquarters.

When engineering a satellite network capable of effectively handling VoIP, there are several important elements that require consideration.

Latency: There is approximately 280 msec of one-way propagation delay due to the location of the Geo Stationary orbit and the speed of light. Regardless of the satellite product, this propagation delay must be considered and overcome. Today, numerous overseas calls originated in the United States are actually transmitted as VoIP over satellite calls, particularly if provided by the smaller long distance carriers.

Jitter: Quantifies the effects of network delays on packets arriving at the receiver. Packets transmitted at equal intervals from the transmitting gateway arrive at the receiving gateway at irregular intervals. Excessive jitter has the effect of making speech choppy and difficult to understand. Jitter is calculated based on the inter-arrival time of successive packets. For high-quality voice, the average inter-arrival time at the receiver should be nearly equal to the inter-packet gaps at the transmitter and the standard deviation should be low. Jitter buffers (packet buffers that hold incoming packets for a specified amount of time) are used to counteract the effects of network fluctuations and create a smooth packet flow at the receiving end.

Packet Loss: Packet loss or packet corruption will cause degradation of voice quality. Since all of the voice traffic is UDP/IP and would not be retransmitted (like in the case of TCP/IP) all packets would be completely lost if the packet becomes lost or corrupted. It is extremely important to have very low Bit Error Rates (BER) to ensure low or no corruption or loss.

QoS and Traffic Prioritization: Packet switched networks are subject to congestion as typical data traffic is bursty. Congested networks can wreak havoc on a VoIP call with delayed , dropped , or packets that are out of sequence . It is a necessity to have QoS and Prioritization in order to guarantee delivery of VoIP traffic through congested links.

Compression Technologies: There are many encoding schemes that have been standardized for voice. The most dominant standard in the industry is the G.729 codec. G.729 encoding requires 8Kbps of bandwidth, but because of the overhead associated with IP/UDP/RTP headers, the actual bandwidth needed is between 16 and 18Kbps (depending on the equipment vendor and configuration). With Compress RTP (cRTP) the total bandwidth requirement per call will drop to about 10Kbps. There are other standards, such as G.723 that takes only 5.3Kbps for the voice payload.

Required bandwidth per VoIP call: To design a network properly, one would need to know the amount of bandwidth required per VoIP call, the number of concurrent calls, and the duration of the call. Bandwidth required per VoIP call will depend the encoding standard used, header compression, and payload size. For example, with G.729(b), payload of 40 bytes, and no header compression, a VoIP call would take about 16Kbps of bandwidth. With header compression this would require 10Kbps of bandwidth.

The iDirect Solution

iDirect technology provides an industry-leading solution allowing enterprises to design networks that simultaneously support data, voice, and video traffic. iDirect delivers unparalleled features and functionality which guarantees VoIP quality even through congested networks.

iDirect's solution is capable of delivering BER better than 10^{-9} which is considered clear channel transmission and is therefore ideally suited for VoIP applications. iDirect provides its customers with a scalable private network solution where the QoS can be predefined and established. QoS can be assigned by traffic type, origination and destination, and can also be assigned a percentage of the bandwidth. In addition, available CIR (Committed Information Rate) capability allows the customer to plan the needed bandwidth in advance to make sure that all of the VoIP calls are of the highest quality.

iDirect's platform supports H.323 or SIP compliant VoIP solutions thus allowing our customers the freedom to choose from multiple equipment vendors.
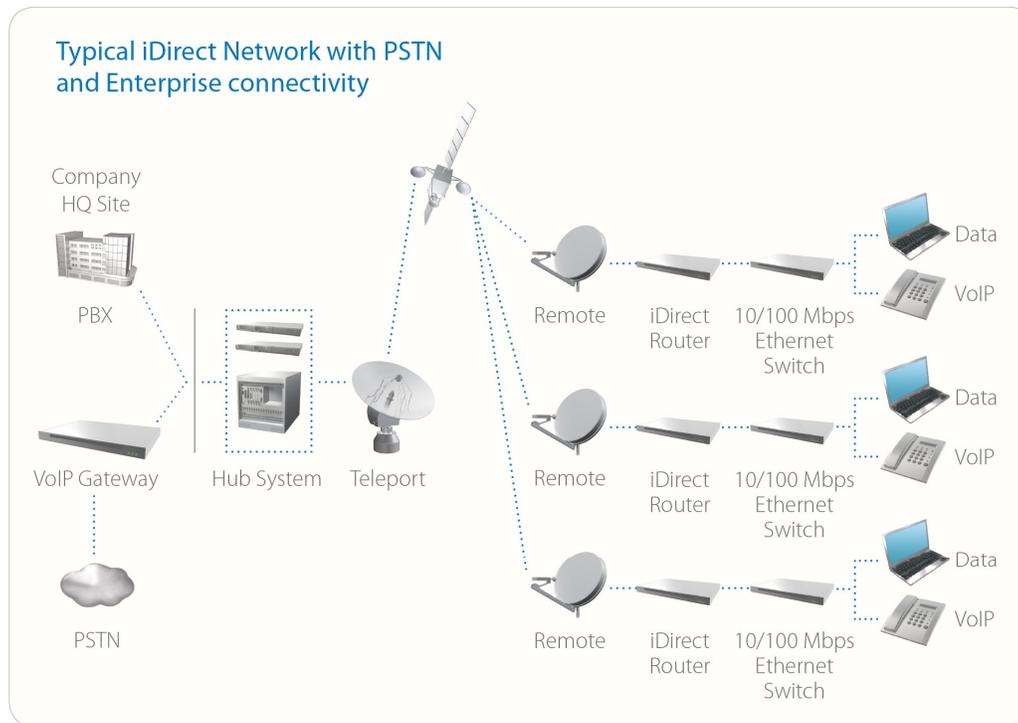

Figure 1: iDirect VoIP Solution

Summary

VoIP technology continues to gain momentum and increasingly carries more traffic as it proves to be a cost-effective method to implement and manage enterprise based voice solutions. With iDirect's differentiated technology the benefits of VoIP are further exacerbated by the tremendous flexibility, availability and reach inherent in a satellite transmission solution.

As with any solution, there are many issues to consider before deploying a VoIP solution. These include latency, reliability, traffic prioritization, QoS, and compatibility. iDirect addresses all of these issues with a cost-effective, technically superior solution.

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