ROI on International Voice Compression

Carriers Squeeze Profits from IPLs with Compression

Recent advances in voice compression technology have created a lucrative market in international telephony services where none existed ten years ago. Historically, there was little profit to be made from leasing and reselling international bandwidth from major carriers.

Limitations in early voice transmission technology made it impossible to carry sufficient voice traffic to justify the carrier's international leased line charges - while making a reasonable profit.

However, with improvements in compression technologies and reductions in price per port charges, it has become viable to replace standard, non-compressed PCM voice services with solutions that minimize the amount of bandwidth required to carry voice traffic at ratios of 8:1, 10:1 12:1 and even 16:1 while maintaining toll-quality voice. On both traditional switched telephony (TDM-based) networks and packet-based architectures (IP) it is now possible to fit enough traffic into rented pipes to offer lower priced voice services to customers, pay the carriers and still collect a hefty profit in the process.

A wide assortment of enterprising International Private Line (IPL) businesses have found a ready-made market for these new technologies among multinational businesses that hope to reduce the expense of backhaul phone services and among temporarily transplanted and immigrant populations seeking affordable ties to friends and relatives back home. One of the newest trends in the market today are pre-packaged, prepaid phone cards (both domestic and international) available on a mass-market basis to consumers. As prices for voice services drop, demand among consumers as well as businesses concerned with economic globalization has skyrocketed. As fast as new IPL entrants are getting into this lucrative business, industry suppliers are cranking out solutions and systems to support them. Making sense of the different algorithms and technologies that are part of voice compression solutions are critical to building the most effective, and profitable, business models.

Figure 1 details the difference in bandwidth required to support the same number of voice channels both with and without new compression technologies. By dramatically cutting the number of leased lines required, thereby reducing the carrier line charges which represent the single biggest operating expense for IPLs - revenue opportunities in this market expand dramatically.

Fig. #1: Comparison of Transmission Line Costs with Voice (PCM) Compression

Monthly leased line costs					
	Circuits	# Voice Channels	US - Brazil	US - China	US - Germany
			Monthly Leased Line Costs		
Without Compression - Traditional PCM - 1 Voice Channel requires 64 kbps	1*E1	30	$50,000	$55,000	$14,000
Large System Without Compression	12*E1	360	$600,000	$660,000	$168,000
12:1 Compression – Veraz I-Gate 4000 series with G.729 codec on TDM bearer	1*E1	360	$50,000	$55,000	$14,000
Savings	11*E1	330	$550,000	$605,000	$154,000
Estimated ROI period for I-Gate			1-2 months	1-2 months	3-5 months

[1]Source: Complete telecom Solutions - June 1999

[2]Estimate based on I-Gate equipment (approx. $500/port)

How Does Compression Work?

From the time carriers began converting PSTN infrastructures to digital networks, engineers have been steadily improving upon the process of converting the traditional voice signal into bits suitable for a data network transmission. Back in the 1960's, the CCITT standardized G.711, better known now as Pulse Code Modulation (PCM), as the internationally accepted coding standard for toll-quality voice transmission. At this standard, a single voice channel requires 64kbps when transmitted over the traditional TDM-based telephony network. The 64 kbps PCM time slot - or payload bit rate -- forms the basic building block for contemporary public telephone services and equipment.

Newer voice compression algorithms try to model PCM more efficiently by using fewer bits. In very simple terms, compression algorithms apply complex mathematical models to the voice stream to reduce the bandwidth required while preserving the quality or audibility of the voice transmission. The Mean Opinion Score (MOS), created by the ITU, has become the accepted measure of voice quality and is determined through a statistical sample of user opinions.

Figure #2 lists a variety of different compression coding standards that have been incorporated into many popular Compressed Voice solutions. Using these standards, the 64 kbps payload bit rate can be reduced to 16 kbps, 8 kbps or even as little as, 6.3 kbps and 5.3 kbps with differences in quality. Significantly, any one of the algorithms can be part of one or more different voice compression solutions. Furthermore, while individual algorithms comply with standards, there is still room for proprietary development and continued enhancements among vendors resulting in even lower rates.

Standard	Name	Payload Bit Rate	MOS[2]
G.711	PCM Pulse Code Modulation	64 kbps	4.4
G.729A	CSACELP Conjugate structure algebraic code excited linear prediction	8 kbps	4.2
G.723	MPMLQ Multipulse maximum likelihood quantization	6.3 kbps	3.9
G.723	ACELP Algebraic code excited linear prediction	5.3 kbps	3.5
GSM-AMR	Adaptive Multi-Rate (*AMR*) voice coder	12.2 kbps	4.14
GSM-EFR	Enhanced Full Rate voice coder	12.2 kbps	3.8

[1] Source: Data Communications, July 1999.

[2] Mean opinion score, MOSs above 4 are considered"toll quality" by the ITU.

Selecting a Voice Transmission Medium

Enhancements to voice compression algorithms have evolved steadily over the past 20 years. These have been deployed over the traditional TDM-based telephony architecture during that time and are the performance proven technology of pioneer IPLs and the carriers themselves.

However, the popularity of the Internet has contributed to an increased interest in VoFR and VoIP protocols among IPLs. There is considerable discussion about the suitability of Frame Relay, IP packet and ATM cell switching technologies for carrying voice traffic.

Since packet technologies were originally designed to handle bursty data traffic, they are inherently less efficient then their TDM-based counterparts in dealing with voice. To achieve good voice quality, the delay of voice packets across the network must be minimal and fixed.

Due to the shared, store and forward, nature of the packet/cell-switching network, it might take time for transmissions to travel across the network. A transmission can be delayed because of network congestion. For example, it might "get stuck" behind a long data transmission that delays other packets. Network congestion can also result in dropped packets, which also detrimentally affects the integrity of voice transmissions.

Figure 3 compares VoIP, which, with the explosion of the public Internet, has become the most popular packet technology with the more traditional TDM-based voice compression solution.

Figure 3: Chart summarizing the leading options (TDM & VoIP)

	TDM (Compressed Voice)	**VoIP (Compressed Voice)**
Technology	Path dedicated to the transmission for the duration of the call, which is sent in a continuous bit stream.	Statistical multiplexing of voice with dynamic allocation of bandwidth based on transmission activity.
Bandwidth Overhead	Overhead can be less than 1%.	Header information needed to address each voice packet adds significant overhead and in some cases can lead to doubling of the bandwidth required.
Delay	Constant bit interleaving leads to low fixed delay from compression algorithm and transmission media.	In addition to the fixed delay, it is necessary to control "jitter", the variable delay resulting from variations in the arrival times between packets.
Quality	Based on algorithm used. Higher quality is typical due to lower delay.	Based on algorithm used, but further degradation is possible due to the responsiveness to silence suppression and lost packets due to network congestion.
Fragmentation	N/A - bit interleaved.	Fragments reduce the overall delay of voice traffic, but adds a lot of overhead due to the large size of IP headers.
Prioritization and QoS	N/A - Dedicated resource ensure highest Quality of Service (QoS).	Network must be properly conditioned to meet VoIP's stringent QoS requirements.
Networking capability	Optimized to replace point- to-point transmission lines.	IP addressing structure provides flexible call redirection services

When comparing transmission technologies to one another, several other points are also worth noting. The TDM bit stream is designed to work best over dedicated terrestrial or satellite lines. In addition, IP voice also work over packet networks.

Other Considerations

While an analysis of transmission alternatives is dominated by highly technical considerations, service providers must also take into account the regulatory environment in emerging and newly deregulated markets. Historically, packet-based technologies were favored over TDM-based solutions: Labeled as data services, Voice Over IP and Voice Over Frame traffic did not fall under the same rules and tariff regulations that applied to the more obvious TDM-based voice compression equipment. Today, as enhanced protocol analyzers offer an easy-to-implement policing tool for regulatory agencies, packet-based traffic can be easier to detect and block than customer-based TDM equipment.

Selecting a Vendor - Enhanced Features

After determining the transmission technology and algorithms that will deliver the right balance to meet bandwidth and voice quality requirements, IPL's still need to wade through a layer of vendor features when evaluating different voice compressed solutions. Some of the more common enhancements include:

- **End to End Compression on Super Tandem:**
  A few vendors, such as Veraz’s I-Gate 4000 series have eliminated the need for compression/decompression on each cascaded link. An embedded intelligent to Super Tandem to remove the cumulative distortion of consecutive compression/decompression cycles.

- **Fax Relay:**
  Most vendors add fax capability with rates configurable up to 9.6 kbps with automatic fallback to lower supported rates across the link.

  I-Gate 4000 series provides fax support on Group 3 fax, ITU T T.38 fax relay or pass-through to G.711.

- **Data Relay:**
  With the emergence of the Internet, modem traffic across international voice circuits is becoming increasingly rare. Therefore modem support is not usually critical when engineering a compressed voice system.

  I-Gate 4000 series provides Voice band data/modem support
  - Pass through to G.711
  - V.22, V.23, V.32, V.34, V.90 and V.92
  - Operator configurable maximum number of VBD/modem call (and transparent channels)

Today, excellent voice quality is possible at rates as low as 4.8kbps. If international link rates are high, the added revenue from dense compression is especially attractive.

Conclusion

Over time, VoIP Packet based transmission format will continue to increase in popularity among carriers who need to converge voice and data. In an IPL environment, where the dominant traffic is voice, the TDM format will continue to deliver higher quality with lower latency and overhead. As bandwidth becomes more abundant, and there is increased interoperability between vendors VoIP systems, we are likely to see an increase in packet based voice compression. Vendors offering a mix of both the TDM and Packet technologies will be able to use TDM for best quality and lowest latency and Packet for flexible switching options.

For those getting into or expanding in the IPL business today, a simple review of the customer's needs will help you identify the best Compressed Voice solution. For simple transmission line replacement and highest quality voice, consider a TDM based voice compression solution.