Maximizing Channel Capacity in a Voice Communications Network

The rapid growth in personal communications has led to a critical need for increased capacity in voice communication networks. This need typically takes the form of a broad requirement to handle more simultaneous voice connections or messages in some constrained bandwidth or bit rate, without degrading power, weight, range or voice quality. The typical solution to this requirement is convert to a digital communication network and to then employ voice compression to reduce the amount of digital data (i.e. bits) which must be handled by this work.

One prominent application where increased capacity has been required is cellular telephony. In this case the large increase in subscribers has over-stretched the capabilities of the original analog cellular systems deployed in the U.S. and around the world. The solution has been a rapid migration to new digital cellular systems where voice compression forms a critical component. Another key example is mobile radio where increased demand from both the public safety and private sectors has led to shortage of spectrum. New digital mobile radio systems such as APCO Project 25 provide a remedy for this problem by using voice compression to increase capacity by a factor of 2-4 times while providing better voice quality and additional features. Another revolutionary example has occurred in mobile satellite communications where voice compression and digital communications has facilitated the introduction by Inmarsat, ICO and IRIDIUM of new, small, go-anywhere telephones which are used by professionals and travelers who want communications at virtually any spot on the globe.

As evidenced by these examples, the combination of voice compression and digital communications is key component in most modern mobile voice communication systems and in many non-mobile systems as well. The basic reason is cost and performance. In a digital system using voice compression the number of users which can be supported in some available bandwidth is greatly increased relative to analog or uncompressed digital systems. The result is that the cost of developing and operating the system can be spread over many more users which typically leads to lower charges per user. Furthermore compressed digital systems can offer better performance in terms of call clarity and features, as well as in terms of terminal size, cost and power.

Voice coding or vocoding is the technology behind most modern voice compression techniques. A vocoder is typically a computer algorithm or program which operates on a digitized voice signal generated by an Analog-to-Digital converter. The vocoder algorithm first encodes a voice signal by processing it in varying way in order to represent it with some small number of bits. A vocoder also contains a decoding function which is able to reconstruct the voice waveform from these bits. Many different vocoder algorithms have been developed which employ different types of processing and depending on the method of processing, some algorithms perform considerable better or worse than others. Vocoder performance is generally measured in terms of compression rate (i.e. how few bits are required to represent the voice signal) plus the voice quality (i.e. how much distortion does the encoding/decoding process introduce into the reconstructed voice signal). Additional performance factors include the complexity of the algorithm, in terms of the amount of computing power required to run the algorithm, and its robustness to factors such as background noise and bit errors which are often present in the real world. Due to these differences, selection of the best vocoder is one of the larger challenges faced by the network designer.

One relatively new vocoding method is the AMBE® Vocoder developed by Digital Voice System, Inc. (DVSI) Burlington, Massachusetts. This method is in the class of Multi-Band Excitation (MBE) algorithms which was initially pioneered at the Massachusetts Institute of Technology (MIT) in the 1980’s. The most notable characteristic of the MBE algorithms is that they employ a sophisticated speech model which decomposes each small section of the voice signal into many frequency bands and then determines pitch, spectral envelope and voice parameters for each section from its frequency components. The IMBE™ and AMBE® Vocoders developed by DVSI are the result of many technological improvements and innovations directed toward the development of a robust, high quality vocoder which can operate at very low bit rates and at relatively modest complexity. While other vocoders such as CELP (Code Excited Linear Prediction) are capable of generating low bit rates they typically introduce a significant loss of quality. In contrast DVSI’s AMBE® Vocoder is the first to demonstrate low bit rates while producing toll-quality speech such as that traditionally associated with wireline telephone systems.

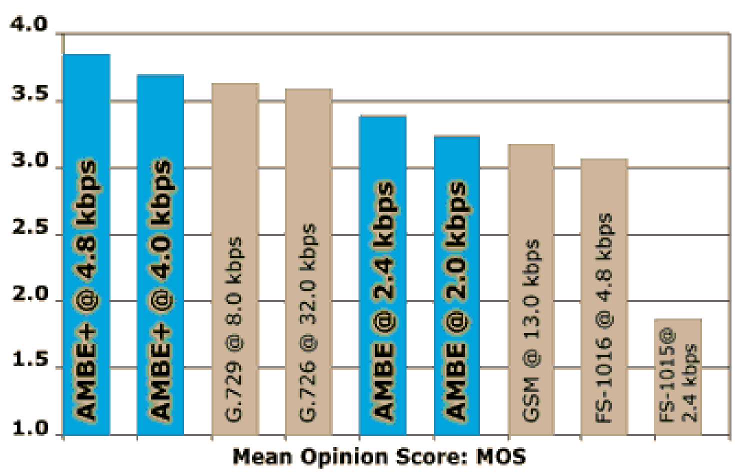

Figure 1: AMBE® Vocoder Performance

Extensive testing has shown that DVSI’s latest AMBE® Vocoder meets the tough performance objectives required for tomorrows generation of communication products. These tests include complexity comparisons, showing it to possess some of the lowest processing power requirements in the industry. In addition voice quality evaluations have established that the AMBE® Vocoder outperforms many standard vocoders operating at several times the bit rate. In fact recent test data compiled by several independent laboratories have provided firm evidence of the AMBE® vocoder's performance advantage. This can seen in Figure 1, where voice quality is represented in terms of Mean Option Score (MOS) with a higher value representing better quality as judged by group of unbiased listeners. Different vocoders, operating at rates between 2 kbps and 32 kbps are shown in this figure. This data shows that the 4 kbps AMBE+™ Vocoder performs as well as the 8 kbps G.729 vocoder and the 32 kbps G.726 vocoder, meaning that a 4 kbps AMBE+™ Vocoder can be used to gain a 2-8 times increase in channel capacity while maintaining the same or even improving voice quality. A similar result holds for the 2 kbps AMBE® vocoder which can be seen to provide better quality than the 4.8 kbps FS-1016 vocoder and the 13 kbps GSM vocoder while yielding a 2.4 - 6.5 times increase in capacity. Based on such convincing evidence, many systems have chosen to use AMBE® or AMBE+™ Vocoder in order to achieve the benefits afforded by such significant capacity gains. These gains include more simultaneous users in a bandwidth limited cellular system, or up to a 16-fold increase in voice traffic capacity with no additional line costs in a wireline system, greater range and quality in mobile radio system, or higher message throughput in a voice paging application. Today the AMBE® Vocoder is a virtual standard in the mobile satellite industry, used by virtually all existing and planned systems. Furthermore DVSI's vocoders are widely used in many mobile radio systems such as APCO Project 25 and they are an ideal candidate for the next generation of digital cellular systems

DVSI has facilitated the widespread use of its AMBE® Vocoder through the introduction of two new products - the MultiMux™ Vocoder Board and the AMBE-2000™ Vocoder Chip. While these products are each focused on a different part of a voice communication system, they are intended to work together to provide a complete vocoder system solution.

The MultiMux™ Vocoder Board ("Best of Show" Teleconnect, November 1998) is ideally suited for communication applications that require toll quality speech. The product includes DVSI's latest 4.0 kbps AMBE+™ Vocoder to maximize channel capacity while providing superior quality to many alternatives operating at over twice the rate. With its unique patent pending design, the MultiMux™ Vocoder Board provides 16 independent channels of compressed speech which makes it ideal for base station or gateway applications where many simultaneous users must be served in an efficient manner. The MultiMux™ Vocoder Board features a dual bus (H.100 and PCI) architecture to allow easy interconnect to a wide variety of interface products including T1/E1 channels as well as ISDN and analog connections.

The AMBE-2000™ Vocoder Chip is an extremely flexible single chip (100 pin TQFP) vocoder running DVSI's AMBE® and AMBE+™ vocoder algorithms. This low -cost integrated circuit can operate in either full-duplex voice or half-duplex modes. It can be configured to operate at virtually any bit rate between 2000 – 9600 bps, and it includes integrated Forward Error Correction (FEC) and Vitterbi decoding to allow operation in the most demanding channel conditions. Other features include integrated echo cancellation, voice activity detection (VAD) and comfort noise generation, DTMF detection and signaling, and low power (3v) operation. The AMBE-2000™ Vocoder Chip is targeted at mobile terminals and portable devices which require voice encoding and decoding in a small high performance package. In addition it is designed to work in conjunction with the MultiMux™ Vocoder Board to provide a complete 4 kbps vocoder solution, where the MultiMux™ operates at the gateway and the AMBE-2000™ operates in the user terminal. Various uses of these products are highlighted in some of the examples presented below.

A wireless local loop application, such as that shown in Figure 2, demonstrates the ability of the MultiMux™ Vocoder Board and the AMBE-2000™ Vocoder Chip to work together to form a complete vocoder solution in a wireless communication application. In addition to wireless local loop, this combination can be readily applied to other wireless voice applications such as voice paging, land mobile radio or GMPCS terminals. In this example application the wireless system is designed around a base station containing one or more MultiMux™ Vocoder Boards. Each board is connected to the Public Switched Telephone Network (PSTN) via a T1/E1 connection to provide the telephone interconnect capability for each mobile user. The MultiMux™ Vocoder Boards internal echo cancellor provides an echo-free telephone connection and the board's high performance AMBE+™ vocoder software enables each board to encode and decode up to 16 simultaneous voice calls at 4 kbps per call. The compressed bits transmitted and received between the base station and each mobile handset using a full-duplex wireless radio link. The very low bit rate per call (4kbps plus optional FEC) allow superior range and robustness to be designed into the link. The AMBE-2000™ Vocoder Chip, built into each handset, provides all necessary vocoder functionality. In addition the chips internal FEC functions can be enabled to improve range and call quality at no extra cost, and its low voltage, low power operation provide for superior talk-time and standby time.

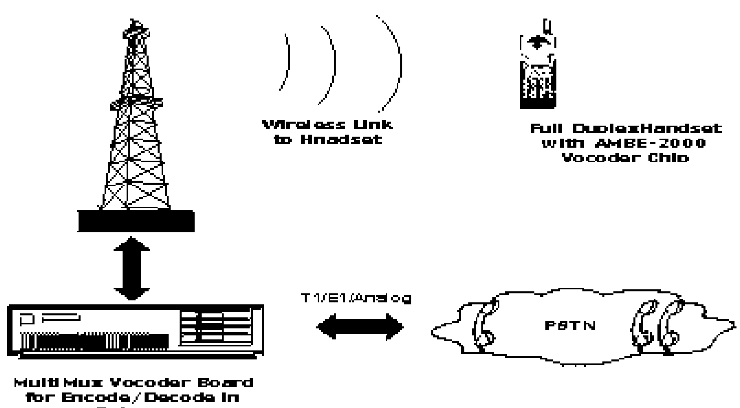

Figure 2: AMBE-2000™ and MultiMux™:
A Complete Vocoder Solution

The MultiMux™ Vocoder Board is also well suited to operate in a gateway only configured such as that pictured in Figure 3. This example depicts a block diagram of a typical satellite based configuration using the MultiMux™ Vocoder Board. Each board compresses sixteen 64 kbps PCM voice channels from a standard T1/E1 interface into a single 64 kbps multiplexed bit stream, however several MultiMux™ boards may be used together to increase call capacity for a larger gateway. In the application being shown the compressed bit stream is sent to a VSAT satellite terminal which serves as an inter-gateway link allowing transmission of the compressed voice bits between virtually any points on the globe.

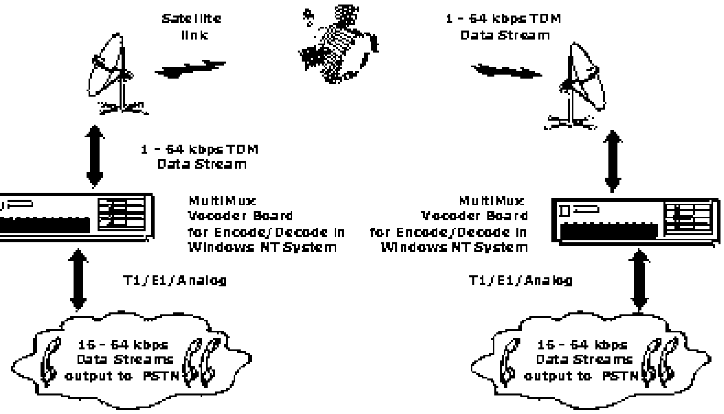

Figure 3: MultiMux™ Vocoder Board:
Satellite Gateway Application

The example shown in Figure 4 represents a similar application where the satellite link has been replaced by an ISDN or T1/E1 connection. In this case the 64 kbps multiplexed data is generated by the MultiMux™ and then routed across an ISDN connection. If additional MultiMux™ boards are used together (each handling 16 independent voice channels) the ISDN connection may prove insufficient in which case it can be replaced with a full or fractional T1/E1 connection.

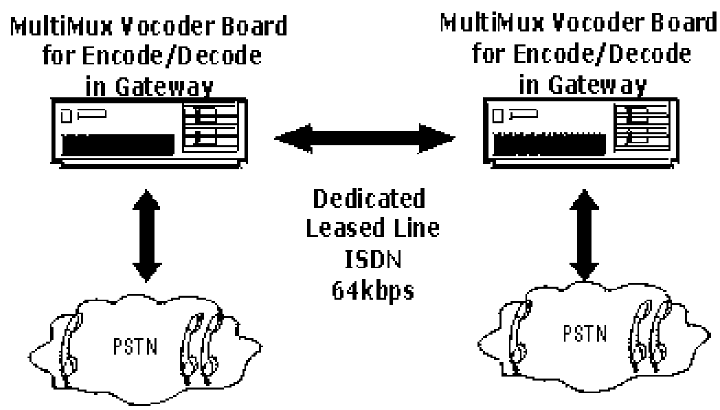

Figure 4: MultiMux™ Vocoder Board:
Wireline Gateway Application

The benefits of using the MultiMux™Vocoder Board and the AMBE-2000™Vocoder Chip in the above applications are numerous. The compressed speech is toll-quality at a very low bit rate of 4 kbps. The lower data rate translates into more available channels or greater transmission range and/or less spectrum usage. And finally, the relatively low complexity of the algorithm results in lower power consumption and longer battery life.

At the heart of these voice network examples resides the AMBE+™Voice Compression algorithm that is used in all major satellite systems to provide toll quality speech at 4.0 kbps. This unique algorithm when combined with the MultiMux™ Vocoder Board provides a superior solution for more effective utilization of bandwidth and channel capacity in voice communication networks.

In summary when network congestion effects the Quality of Service seek the solution that will maximize bandwidth potential without sacrificing speech quality. Digital Voice Systems, Inc. will provide the solution that provides toll-quality speech while will maximizing capacity to mitigate the impact of voice network congestion.

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