SPEECH CODING AND ADAPTIVE CHANNEL CODING TECHNIQUE FOR MEMORY AND MEMORYLESS CHANNELS

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ABSTRACT

The speech and audio compression that have emerged are cost effective with diverse commercial applications. The services offered by the mobile communication [12] system are enormous and hence number of mobile subscribers are increasing at a larger rate. To accommodate all these users we need to redesign the cellular system which is a tedious process. We can overcome this draw back by using the speech compression technique with adaptive channel coding technique. This paper aims at designing an algorithm for memory and memory channels with high S/N ratio. Here the noise effect of the channel is predicted by transmitting a hand shake packet between the transmitter and the receiver. If the channel is erroneous then we use full rate packet coding else we use half rate packet coding hence increasing the transmission efficiency. The coded data is transmitted using the QAM technique.

Key words: LD_CELP, QAM, CYCLIC CODER, CONVOLUTION CODER.

I. INTRODUCTION

The analog I/P speech is converted into the digital data by using the A/D converter. We use two sampling frequencies. i.e. 8Khz and 6Khz. The output digital data rates obtained after sampling are 64kb/sec and 48kb/sec. If this digital data is transmitted directly then we need maximum bandwidth. To overcome this draw back we are using a speech compression technique. The speech compression technique used is LD_CELP. By using this technique the speech is compressed at the rate less than 16kb/sec. The compressed data is further encoded using the channel coding techniques. The channel coding techniques chosen mainly depends on the S/N of the handshake packet received. If the S/N ratio of the packet is high then we use cyclic coding technique, where number of redundancy bits added for error correction and detection will be less else we use convolution coding technique with large number of redundancy bits added to protect the data transmitted. The channel encoded data is transmitted using the QAM technique.

II. SPEECH COMPRESSION TECHNIQUE

The speech signal obtained will be sampled at the rate two different sampling rates and the digital data obtained after sampling will be 64kb/sec or 48Kb/sec. The total bandwidth available for mobile communication is very less hence to accommodate maximum number of users we use speech compression technique. The speech compression technique proposed is LD_CELP algorithm. This algorithm compresses the speech by less than 4:1 ratio. LD_CELP [1] uses backward adaptive linear predictor which reduces the coding delay. The processing time required in LD_CELP is less than 2 msec and the quality of the speech is similar to the of PCM coded speech. In a backward adaptive configuration, the parameters of the synthesis filter are not derived from the original speech signal but are computed by backward adaptation method. The information is extracted only from the reconstructed signal based on the transmitted excitation information. Since both the encoder and decoder have access to the past reconstructed signal, side information is no longer needed for the synthesis filter and hence the low processing delay requirement is met [2].

In Abs-LPC coding system a closed loop optimization procedure used to determine the excitation signal which when used to excite the model filter produces a perceptually optimum synthesis speech signal. The basic idea behind Abs is as follows. First it is assumed that the signal can be observed & represented in some form i.e. time or frequency domain. The model has a number of parameters which can be varied to produce different range of the observable signals. To derive the representation of the model a trial & error procedure is applied. In the LD-CELP only the excitation signal is transmitted. The predictor coefficients are updated by performing a LPC analysis on the previously quantized speech. Thus the LD-CELP coder is basically a backward adaptive version of the conventional CELP coder.

In this algorithm only the index to the excitation codebook is transmitted. The predictor coefficients are updated through the LPC analysis of previously quantized speech. The excitation gain is updated by using the gain information embedded in the previously quantized excitation vector. The block size for the excitation vector & gain adaption is five samples only. At the LD-CELP encoder as shown in the fig: 1 the encoder I/P digital signal is partitioned in to block of five consecutive I/P signal sample. For each I/P block the encoder passes each of 1024 candidate codebook vector through a gain scaling unit & a synthesis filter. From the resulting 1024 candidate the encoder identifies the one which minimizes a frequency weighted mean square error measure with respect to the I/P signal vector. The 10 bit codebook index of the correspondent best codebook vector which gives rise to the best

candidate quantized signed vector is transmitted to the decoder. The best code vector is given to gain scaling unit & the synthesis filter to establish the correct filter memory in preparation for the encoding of next signal vector. The synthesis filter coefficients & the gain are updated periodically in a backward adaptive manner based on the previously quantized signal & gain scaled excitation.

At the LD_CELP decoder as shown in fig: 2 the decoding operation is preformed on block by block basis. Upon receiving each 10 bit index the decoder performs a table look up to extract the corresponding code vector from excitation codebook. The extracted code vector is then passed through gain scaling unit & a synthesis filter to produce the correct decoded signal vector. The synthesis filter coefficients & the gain are updated in the same way as in the encoder. The decoder signal vector is then passed through an adaptive post filter to enhance the perceptual quality. The post filter Coefficients are updated periodically using information available at the decoder. In LD_CELP the excitation vector has a dimension of five samples. The long term predictor in conventional CELP has been replaced by a high order STP predictor whose coefficients are updated once for every four excitation vector by using a 10th order adaptive linear predictor in the logarithmic domain. The coefficients of the log gain predictor is updated once every four excitation vectors by performing the LPC analysis on the log gain quantized coefficients & scaled excitation vector. At 16 kb/sec only 10 bits are available from the quantization of five samples. The excitation vector quantization codebook is made up of a 3 bit gain & 7 bit shape codebook.

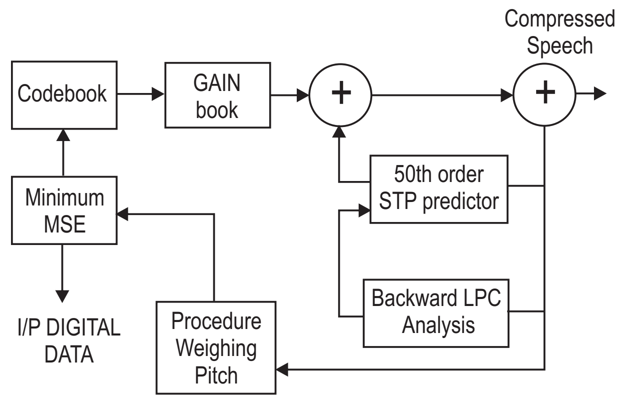

Fig. 1. LD_CELP ENCODER

The O/P data rate from the LD_CELP encoder if the signal is sampled at the rate of 8 KHz will be less than 16 Kb/sec, if sampled at 6 KHz the data rate obtained will be 12 Kb/sec. This obtained data is further given for the channel encoder. In the channel encoder the error detection and correction redundancy bits are added at the transmitter. The amount of redundancy bits that are to be added at the transmitter is measured by the ratio of the number of information symbols in the message to that in the code word.

III. CHANNEL CODING TECHNIQUES

The compressed data from the speech coder is further given to the adaptive channel encoder. Channel coding [16, 10] is a viable method to reduce information rate through the channel and increase reliability. This goal is achieved by adding redundancy to the information symbol vector. The resulting symbol vector is a longer coded vector of symbols that are distinguishable at the output of the channel. The purpose of forward error correction (FEC) is to improve the capacity of a channel by adding carefully designed redundant information to the data being transmitted through the channel. The process of adding the redundant information is known as channel coding. In this paper we are using adaptive channel coding technique. Depending on the quality of the channel the channel coder will be decided. The channel coders used in this paper are cyclic coder and the convolution coder. The communication line between the transmitter and the receiver is set to check the noise quality of the channel. The quality of the channel is checked by transmitting the handshake packet which is known by both the transmitter and the receiver. If the S/N ratio of the received packet is more than the threshold then the cyclic coding technique will be used to generate the redundancy bits. The packet size at the output of the cyclic coder will be 10250 bits. Hence cyclic coder generates the half rate speech coded data when the channel is error less or less erroneous. By this methodology we can accommodate two users within one full rate traffic channel. Hence the channel efficiency is improved.

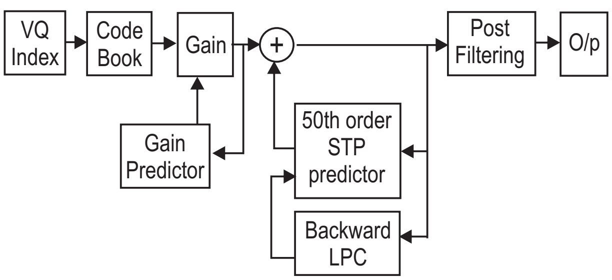

Fig. 2. LD-CELP DECODER

If the S/N ratio of the hand shake packet received is less than the threshold then convolution coding technique will be used. In convolution coding the number of error correcting bits added will be more and it can detect and correcting maximum number of errors. This in turn improves the signal quality through erroneous channel by improving the S/N. The size of the packet at the o/p of the convolution coder is 20500. This is further transmitted by using one full rate traffic channel by improving the S/N ratio.

At the receiver the size of the packet is checked. If the packet size is equal to that of the half rate speech coder then the channel decoder used will be cyclic coder else convolution coder will be used.

IV. MODULATION TECHNIQUE

The channel coded data is modulated using QAM [9] technique. Quadrature amplitude modulation (QAM) is a modulation scheme in which two sinusoidal carriers, one exactly 90 degrees out of phase with respect to the other, are used to transmit data over a given physical channel. Because the orthogonal carriers occupy the same

frequency band and differ by a 90 degree phase shift, each can be modulated independently, transmitted over the same frequency band, and separated by demodulation at the receiver. For a given available bandwidth, QAM enables data transmission at twice the rate of standard pulse amplitude modulation (PAM) without any degradation in the bit error rate (BER). QAM and its derivatives are used in both mobile radio and satellite communication systems.

V. RESULTS

The table below gives the comparative processing time required to compress the speech.

I/P Speech Sample Rate	Rate of Compressed O/P Speech	Processing Time Required
48 Kb/sec	12 Kb/sec	0.000216 sec
64 Kb/sec	16 Kb/sec	0.000242 sec
96 Kb/sec	24 Kb/sec	0.000248 sec
128 Kb/sec	32 Kb/sec	0.0003 sec

The figures below in red color represent the input speech signal before passing through the speech processor and the channel encoder. The figures in green represents obtained decompressed speech at the O/P of the speech processor and the channel decoder. The figure 3 represents the I/p speech at 48 Kb/sec and decompressed speech at the o/p.

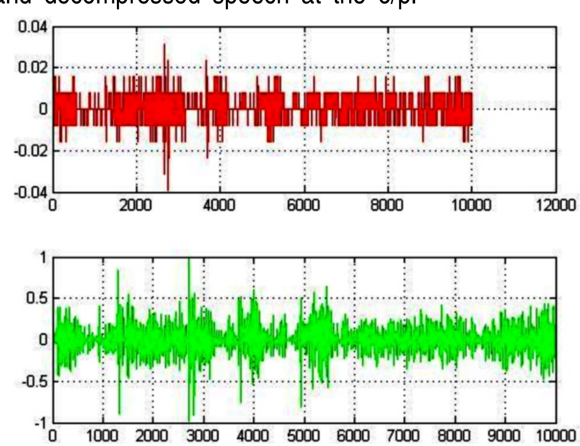

Fig. 3. I/P speech and decompressed speech

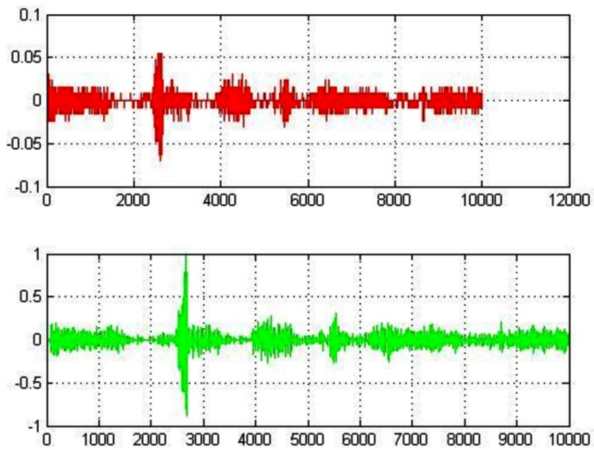

Fig. 4. I/p speech and decompressed speech

The figure 4 represents the I/p speech at 64 Kb/sec and decompressed speech at the o/p

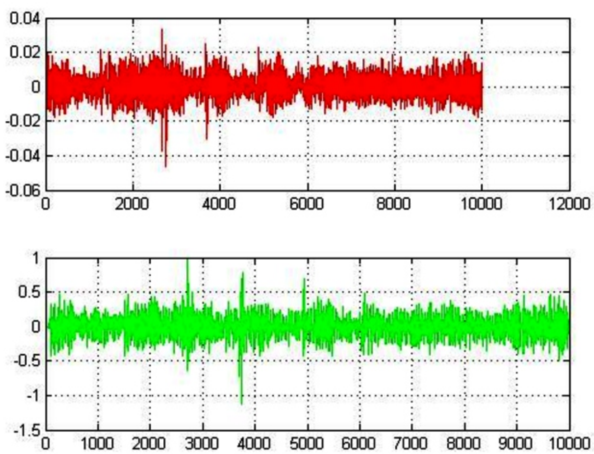

Fig. 5. I/p speech and decompressed speech

The figure 5 represents the I/p speech at 96 Kb/sec and decompressed speech at the o/p

VI. CONCLUSION

The above presented speech compression technique gives toll quality speech signal at 16 Kb/sec with low computation delay. The above presented technique can generate either the full rate traffic data or half rate traffic data depending on the noise level of the channel. If the channel is less erroneous then the coder uses cyclic coder and generates half rate traffic channel hence accommodating more number of user with high S/N ratio. If the channel is noisy then it uses convolution code and generates full rate traffic channel with high S/N ratio with minimum computational delay.

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REFERENCES

[1] Juin-Hwey Chen, Senior Member, IEEE, Richard V. Cox, Fellow, IEEE, Yen-Chun Lin, Nikil Jayant, Fellow, IEEE, and Melvin J. Melchner "A low-delay celp coder for the ccitt 16 kb/s speech coding standard" IEEE journal on selected areas in communications, vol. 10, no. 5, June 1992

[2] Vladimir Cuperman "Low Delay Speech Coding" Communications Science Laboratory, School of Engineering Science Simon Fraser University, Burnaby, B.C., Canada V5A 1S6

[3] A.M. Kondaz, "Digital speech coding for low bit rate communication systems", John Willy & Sons 1999.

[4] ITU-T standards – G. 728 "Coding of speech at 16 Kbits/sec using low delay code excited linear prediction".

[5] ITU-T standards – G.721. "Adaptive predictive coding of speech signals", CCITT Red Book recommendation.

[6] W.H. Kim and C.V. Freiman "Department of Electrical Engineering Columbia University" Multi-error correcting codes for a binary asymmetric channel

[7] Sham Shanmugam "Analog & Digital Communication "

[8] Simon Haykin "Digital Communication"

[9] P.S. Satyanarayana. "Concept of Information theory & coding"

[10] Theodre. S. Rappaport. "Wireless Communication"

[11] Kamilo Feher "Wireless Digital Communication"

[12] IEEE Transaction on information theory Volume 54 No. 1 Jan 2008 "Explicit Codes Achieving List Decoding Capacity: Error-Correction with Optimal Redundancy"

[13] Shu Lin & Daniel. J. Costello "Error Control Coding"

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