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PROPOSED SOURCE AND CHANNEL CODING TECHNIQUES FOR MEMORY CHANNELS

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ABSTRACT

The desired trend in the design of the communication system depends on efficient use of available bandwidth and to improve the s/n. This paper aims at designing source and channel coding algorithms for both erroneous and errorless channels. The noise effect of the channel is predicted by transmitting the packet of data known to both transmitter and receiver in the form of hand shaking. The obtained handshake packet is compared with fixed threshold noise level by the designer and depending on the effect either complex channel coding algorithm or simple channel algorithm is chosen, hence reducing computation complexity. This improves the s/n ratio [8,9].The channel coded data is modulated using QAM technique. The number of levels in QAM depends on the optimal power and bandwidth available .The proposed algorithm includes methodology to improve s/n ratio with minimum computational complexities and high accuracy.

***Keywords:** source coder, channel coder, LPC-linear predictive code, STP -short term predictor.*

1. Introduction:

In the reference paper [17] the coding technique used was joint coding technique .In this method the error correcting algorithm is combined with the source coding technique and in this method we need to have prior information about the channel.
The information about the channel is the information about the noise level of the channel, fading effect of the channel. Hence is not suitable for memory channels. This proposed paper is suitable for memory channels .In this paper we use the LD _CELP source coding technique and two error correcting techniques..The type of error coding technique used depends on the error level of the channel .The error level of the channel is checked by transmitting a hand shake packet between the transmitter and the receiver.
The hand shake packet received indicates the type of the channel. If the packet is erroneous then the channel is noisy hence we use complex channel coding algorithm. The complex algorithm used is trellis encoding algorithm. If the channel is errorless then we use cyclic code technique to en code the data. The encoded data is given to the QAM modulator [15] .The number of levels chosen in the QAM modulator depends on the hand shake received.

THE FLOW OF THE PAPER IS AS FOLLOWS:
1. Introduction.
2. Explaination about A/D and D/A conversion.
3. Explaination about source coding techniques used.
4. Explaination about channel coding techniques used.
5. Choosing QAM technique
6. Flow diagram
7. Results.
8. Conclusion.

2. Explaination about A/D and D/A conversion:

The continuous speech signal is converted to digital speech using TLC 320 AD532 codec chip. This codec can be programmed for both A/D & D/A conversion [6,7]. As digitally encoded speech ultimately condenses down the binary sequences all the advantages offered by digital system are available for exploitation but disadvantage of the digital system is that it needs extra band width. This can be overcome by using compression technique.

3. Explaination about Source coding techniques used:

The LD-CELP mainly works on the principle of Abs LPC coding system. 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 either the time domain or in frequency domain. The

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represented model has a number of parameters which can be varied to produce different ranges of the observable signal. To derive the representation of the model a trial & error procedure is applied.

In forward adaptive CELP coder the predictor parameters together with the excitation vectors are all transmitted to the receiver. 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. Here scaling unit & the synthesis filter are updated to establish the correct filter memory in preparation for the encoding the next vector

A.COMPRESSION & DECOMPRESSION METHODOLOGY USED BY LD_CELP

LD_CELP is an analysis by synthesis approach for codebook search Similar to that of CELP. The LD_CELP [2] algorithm uses backward adaption of predictor coefficients & gain coefficients. 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.

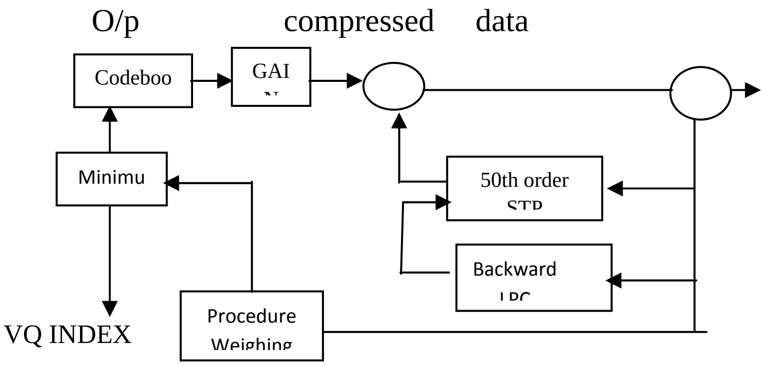


Figure (1) LD_CELP ENCODER

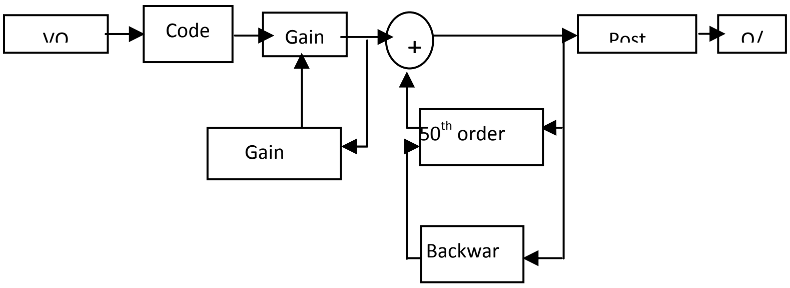
At the source encoder as shown in the figure (1) I/P digital signal is partitioned in to block of five consecutive I/P signal sample. For each I/P block the encoder passes each of 1024 candidate codebook vector through a gain scaling unit & a synthesis filter. From the resulting 1024 candidates the encoder 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 corresponding best codebook vector which gives rise to the best candidate quantized signed vector is transmitted to the decoder. The best code vector is given for the gain update of the next signal vector. The synthesis filter & the gain coefficients are updated periodically in a backward adaptive manner based on the previously quantized signal & gain scaled excitation**.**

At the LD_CELP [1] decoder as shown in the figure(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 [3] to enhance the perceptual quality. The post filter Coefficients are updated periodically using information available at the decoder. Here 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 pole for every four excitation vector by using a 10^{th} 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 time gain quantized coefficients & scaled excitation vector coefficients. At the output of the LD_CELP encoder as shown in the figure 1 we get only 16kb/sec compressed speech and the each vector output after compression is represented by 10bits. The excitation vector quantization codebook is made up of a 3 bit gain & 7 bit shape codebook. Only 16Kbits of data is transmitted between the transmitter and receiver for every 64kbits of data .Hence to protect the data we use channel coding technique .The channel coding technique used help us to detect the error and to correct the error in the data. For error detection error and correction the redundancy bits are added at the transmitter.

4. Explaination about Channel coding techniques used:

For error correction [13] the redundancy bits are added at the transmitter. The amount of redundancy bits [12] that are to be added at the transmitter is measured by the ratio of the number of information symbols in the message to that of in the codeword. The number of redundancy bits are equal to

[k/n]= [message bits/code bits]



Figure(2) LD-CELP DECODER

Channel coding [5] 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 resulting in 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 some carefully designed redundant information to the data being transmitted through the channel. The process of adding this redundant information is known as channel coding. Depending on quality of received hand shake packet two channel coding techniques are used. Different coding techniques are used depending on noise level of the channel

1] If the channel noiseless or if the noise level is less than the threshold then we use cyclic coding technique.

2] If the noise level of the channel is greater or equal to the threshold then we use convolutional coding technique.

If the received handshake packet is errorless then we use cyclic coding [4] technique to generate the redundancy bits for error detection and correction .The cyclic code word c(x) is generated using the equation given below.

c (x) = m(x)g(x).

At the channel decoder if received vector is the function of transmitted vector C(x) and noise present in the channel
e(x) and can be written in the form of equation given below.

V(x) = m(x) g(x) + e(x)

Where V(x) is the received code vector
e (x) is the error vector.
Cyclic codes can be used to correct errors, like single error, double errors and burst errors. Cyclic codes can also be used to correct double errors over the field *GF* (2). The received codeword can be corrected

V(x)mod g (x)=[a(x)g(x)+e(x)]modg(x)

The advantage of using cyclic code is simple to design and channel efficiency is increased as number of error correcting bits added are less.

If received handshake packet is erroneous then we use convolutional encoding algorithm and viterbi decoding algorithm. If the channel is erroneous then we use convolutional coder to encode the data .The advantage of using this encoder is that it can correct and detect any number of errors but the disadvantage of method is that the band width required is more. At the channel decoder we use here is viterbi decoding algorithm.

5. Choosing QAM Technique:

The channel encoded[4] data is further given to the QAM modulator .The number of levels of the QAM modulator depends on the noise factor of the channel used .If the channel is erroneous the number of levels chosen will be less. Hence the available voltage will allocated to selected number of levels. Hence each level will have sufficient power which minimizes the effect of noise. But the disadvantage of reduced number of

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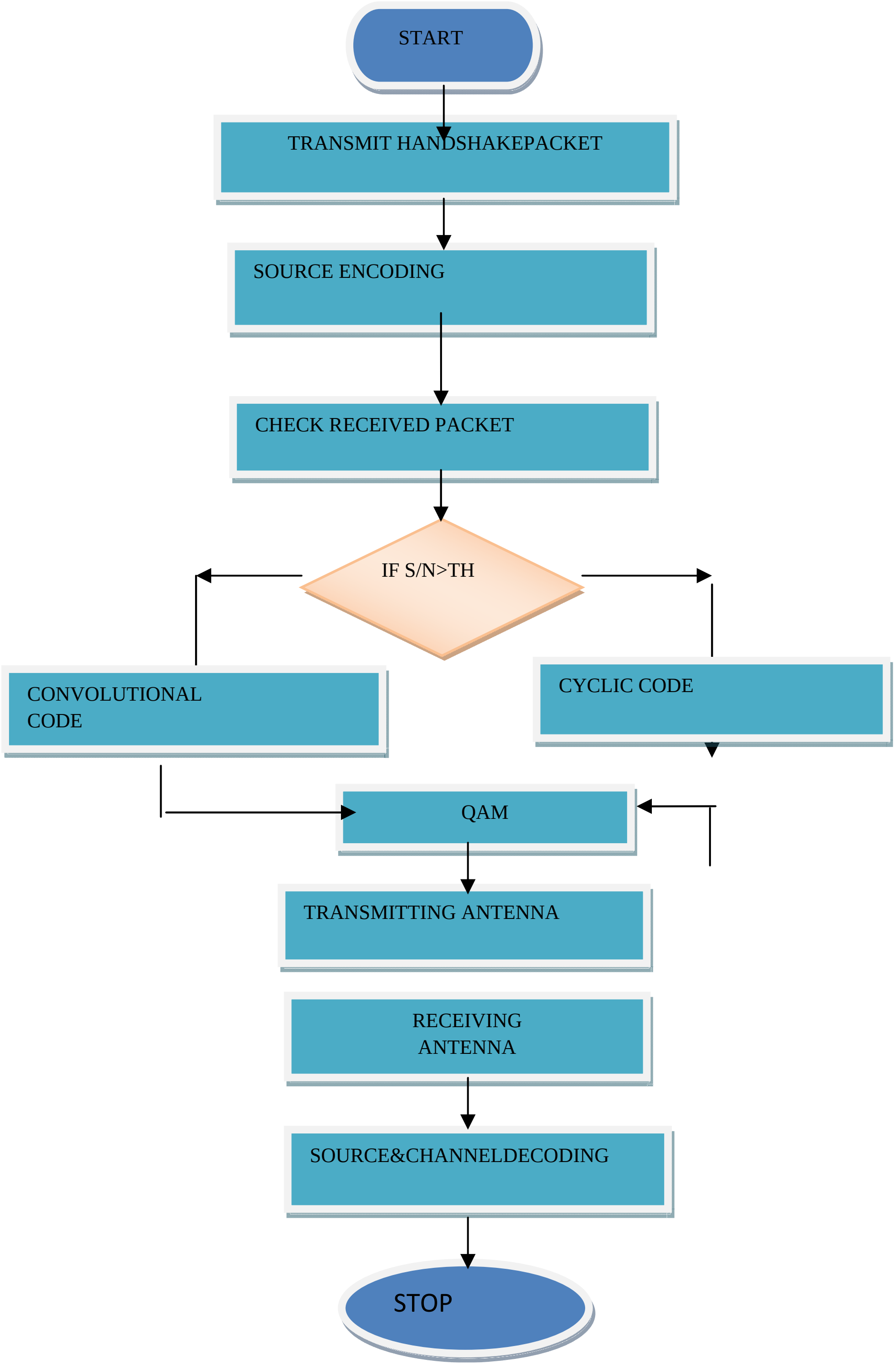
levels is the reduced channel efficiency. If the channel is noise less then number of levels chosen will be high hence increasing the channel efficiency.

6. Flow diagram



7. The Obtained Results after Compression and Decompression

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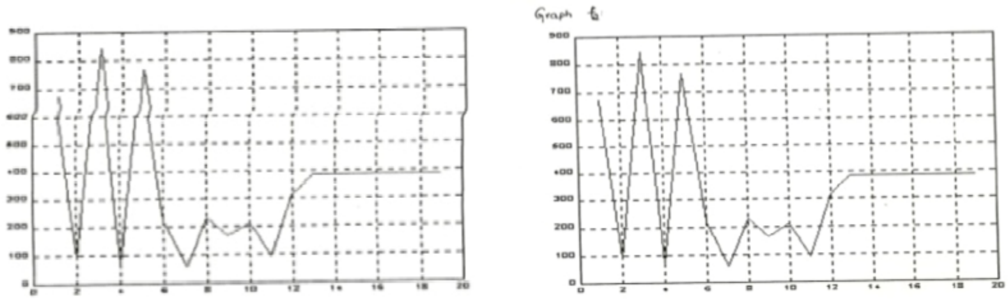

Figure (3) Compression & Decompression results obtained

8. Conclusion:

The proposed method improves the channel capacity by compressing the speech with least delay and selects the channel coding technique depending on quality of received handshake hence reducing the computational complexity without compromising with the quality of signal.

9. Acknowledgment:

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