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Performance Analysis of different Discrete Time and Discrete Amplitude Representation Techniques used for Encoding the Speech Waveform Digitally

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Abstract—This study is mainly based on comparison between different source coding techniques on the basis of their signal to quantization noise ratio.On the basis of the observation some perceptual and finer adjustments were done to find optimized parameters for different techniques.

The main problem for a uniformly quantizer was its deteriorated performance when the amplitude of signal exceeds the dynamic range of quantizer which in turn leads to signal dependent noise components leading to less SNR. So to rectify this we go for compression of signal that is to be presented to the quantizer. Also to increase the performance we introduced a concept of adaption in step size of the quantizer which is based on the input signal.

Along with this, two bandwidth reducing techniques namely differential pulse code modulation (DPCM) and Adaptive Pulse code modulation (ADPCM) techniques were compared on the basis of Signal to noise ratio and it was found that for low signaling rate ADPCM was best but after an instant Pulse code modulation technique outperforms the other three with respect to signal to noise ratio.

Lastly ,we found that when we observe the coding techniques involving one bit quantizer the SNR was largely depending on the amount of fluctuations in input speech signal for optimized delta modulation scheme whereas in case of adaptive delta modulation scheme it mainly depended on the initial amplitude of speech signal.

1. Introduction

Nowadays there are large number of telecommunication and mobile communication network that exist in the world. One of the most important thing that is required is co existence and transmission of so many signal efficiently and occupying less channel bandwidth. In order to do so we need to transmit data having less signaling rate along with high signal to noise ratio. Here comes the role of various source coding techniques .

Source coding techniques basically deals with efficient representation of input analog signal to a digital domain specification. The main work of source coding techniques is to represent a signal in minimum number of bits possible.

The main aim of this paper is to find the values of optimized parameters for different source coding techniques so that they can be used for efficient transmission of information. To accomplish this task we have used a real time speech signal which was given to the system via microphone and then we processed the data and provided it to the inputs of various source coding techniques. The encoding techniques that we are going to implement are basically straightforward approximation of input speech signal and representing them into discrete amplitude and discrete time signal. To deal with each of the encoding techniques we have divided this paper into six sections. Dealing with every aspect of speech coding techniques theoretically avoiding much mathematical digression

Section II deals with different aspects of Pulse code modulation (PCM). Inthis section we will try to compare PCM technique implemented using different types of quantization scheme namely uniform and non-uniform quantization. Afterwards we will see that when the input speech is less in amplitude as compared to the step size of the uniform quantizer there will a huge amount of loss in signal at lower amplitude, So using non uniform how can we eliminate such ambiguities.

Furthermore in section III we will see thathow by using adaptive quantizer instead of normal uniform quantizer we can improve the SNR to a huge extent.

In section IV we will be mainly focusing on differential pulse code modulation technique (DPCM). This techniquebasically deals with encoding the difference of predicted and input speech signal rather than whole thereby reducing number of bits required to quantize a signal , therefore dependency of signal on bit rate reduces and we will see that how there is a variation in SNR if the value of prediction coefficient changes.

Next section comprises an adaptive approach towards differential encoding techniques. Here we will first find the optimized parameter corresponding to the input speech signal and then quantize the differential with those optimized parameters. Then a plot of SNR versus bit rate will be dealt.

In section V&VI we will be dealing with 1 bit quantization scheme i.e. linear delta modulation and adaptive linear delta modulation. In this section our main focus will be on the variation in SNR with respect to the sampling frequency that we are going to use .For linear delta modulation scheme we will find the value of optimized step size . Along with this we will be seeing the distortions such as slope overload and granular distortion and how they can be eliminated by the use of adaptive delta modulation.

In entire paper entire observations were done by considering a speech signal “Apples are red and grapes are green” with a sampling rate of 32000 samples in 4 seconds.

2. Pulse Code Modulation

Pulse Code Modulation(PCM) is a source encoding technique that mainly deals with approximation of a given signal to its nearest discrete amplitude value. [1]Firstly the data is sampled and is provided to a quantizer which has a task of rounding of the input to its nearest possible discretized amplitude value .Then the quantized value is encoded in bit zero and one.Herewe will be dealing with quantization noise which mainly occurs due to approximations done by quantizer.

For performing a Pulse code modulation we have to follow certain procedure:

1) Data should be sampled at a rate greater than Nyquist rate.
2) For uniform quantization the input signal should lie within the dynamic range of the quantizer otherwise there will be a signal dependent error component.Also number of bits required to quantize a signal should be greater than 6 to reduce the signal loss at lower amplitude .
3) The input signal should be passed through a low pass filter before sampling is done to remove unwanted noise signals.[2]

Here we will be considering two scenarios one with respect to a sinusoidal signal in figure a1 and other with respect to a real time speech signal.

While considering a sinusoidal signal of magnitudepeak to peak 2 units and a quantizer with dynamic range of 4 volts we found that the error that we were getting was very random and can be considered as additive white guassian noise but in the case when the range of quantizer decreases the error starts depending on the signal(c) which is not required at all.It also leads to decrement in value of SNR at a very large extent.

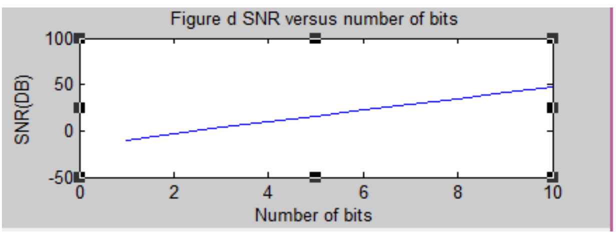

Figure a0.Plot between SNR and number of quantization bits

To deal with such problem one thing that we can do without affecting the design of quantizer is adding a pseudorandom noise also called dither to the output error function thus making it random and after reconstruction of the signal it is possible that you will get an exact signal[3][4]

Now if we consider that all requirements corresponding to PCM with uniform quantization are fulfilled than we can consider the quantization noise uniformly distributed [7]

Hence probability of error is given by: P(e)

P(e)=(1/2d)where(-1/d)<e<(1/d); (1)

Where d is the step size.

Correspondingly,

$$SNR= (X_{rms})^2/(d^2/12); \quad (2)$$

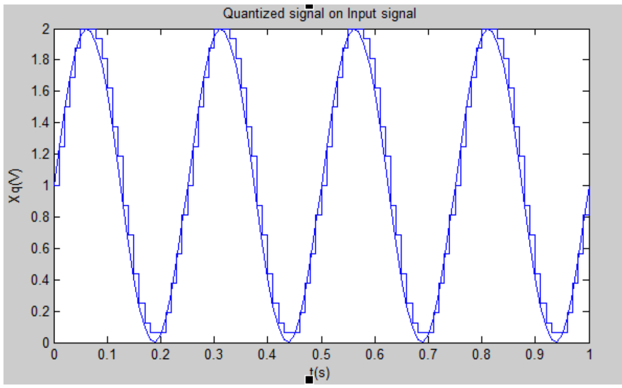

Fig a1. Input sinusoidal signal with Quantized output

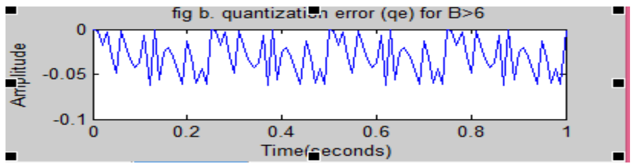

Fig b. Quantization error when input is within dynamic range of Quantizer

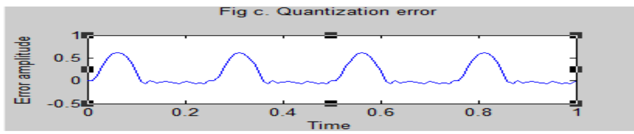

Fig c. Quantization error when input exceeds dynamic range of Quantizer

From the graph a0 it is quite visible that as the number of quantization levels increases there is a drastic increase in signal to noise ratio and hence the dependency of error on signal reduces.

There is one more problem associated with uniform quantization that is for signals with very less amplitude there is a huge attenuation. Todeal with thisProblem we introduce concept of non uniform quantization where we will increase the quantization steps from the origin linearly.This can be done in other way also by compressing the input signal by Smith[6]for logarithmic PCM quantization according to which we can bring all low amplitude signal to higher value when providing it to uniform quantizer(figure1.2) .

$$|v|=|V|\log(1+u|X|/V)/\log(1+u) \quad (3)$$

u>0;

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After quantization we can expand the signal to retrieve the required quantized output. In this quantization scheme the error does not depend on the characteristics of the quantizer and hence there won't be any change if we are going to modify the input signal with amplitude larger than the dynamic range of quantizer. Characteristics of equation (3) is given in figure 1.1.

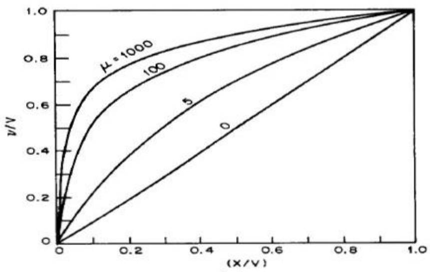


Fig 1.1 Logarithmic compander characteristics

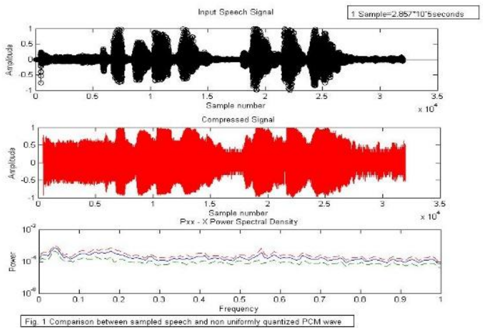


Fig 1.2 Analysis of PCM scheme with non-uniform quantizer

From figure 1.2 one thing is clear that after compression of the signal the output signal becomes uniform in amplitude. The main advantage of this is all the low amplitude signals can be quantized with the given dynamic range of uniform quantizer.

But this comes at a cost of highly fluctuating power of error at lesser frequencies at which we are going to transmit our signal.

This is a figure of demerit for the signal having less frequencies.

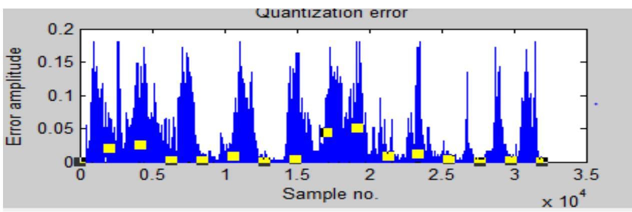


Figure 1.3 Quantization error of non-uniform PCM scheme

Correspondingly as we have mentioned earlier that since by compression of signal the signals of less amplitude are also quantized, so in Figure 1.3 we see that there is a clear reduction in quantization error signal as compared to Uniform PCM scheme. The maximum error that can be observed is of magnitude 0.16-0.17.

Last but the most important factor is SNR. As we can see in figure 2 that there is a linear relationship between SNR and number of quantization bits that are required also reduces as compared to PCM signal with a Uniform Quantizer. As we know that there is a direct relationship between signaling rate and Quantization bits if sampling frequency is kept constant therefore Signaling rate is reduced hence a large amount of bandwidth is prevented.

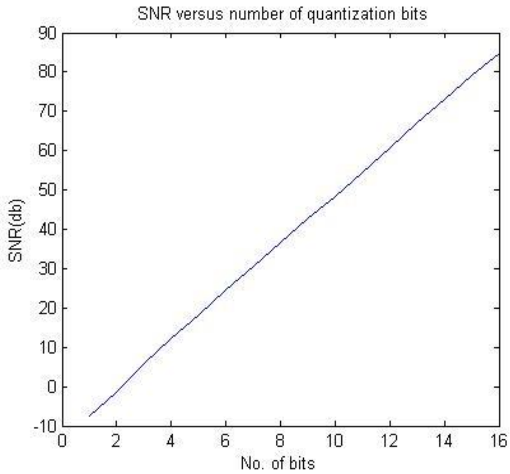
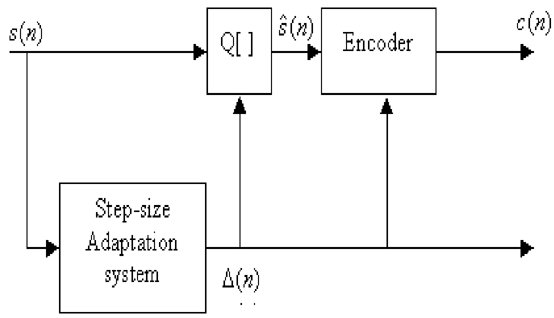


Fig2 Relationship between SNR and number of quantization bits for non uniform quantization

3. Adaptive pulse code modulation

We know that the speech signal is highly non stationary in nature hence if we try to implement a uniform quantization there would be high amount of noise due to highly correlated adjacent samples. Hence we can harness this property of Speech by the use of adaptive quantizer which can predict next step size base on the previous one.

In adaptive pulse code modulation scheme there is only one difference, that is the use of adaptive quantizer instead of non-uniform quantizer that was used previously.The main reason of using adaptive quantization is that in this we are going to decide the step size of next quantization level on the basis ofknowledge of previous quantization step size.This scheme adapts to the changes in input speech signal and hence probability of error decreases.



Feed-forward adaptive quantizer coder

Here we are going to implement uniform quantizer by suggestions made byCumminskey[8]according to which

$\hat{s}_{(n)}=H_r*\Delta_r/2;$ (4)

$H_r=\mp 1,3,5,\ldots 2^b-1$ (5)

Where b is number of bits required for quantization.

$\Delta_{r+1}=\Delta_r*M(|H_r|)$ (6)

$M(|H_r|)$ is step size multiplier. Note that the value of multiplier depends on the latest decoder output. The step size multipliers were selected to maximize the value of SNR

STEP SIZE MULTIPLIER TABLE

Table1 : This table represents value of multipliers M for maximum SNR[7]

CODER	PCM			DPCM		
B	2	3	4	2	3	4
M_1	0.60	0.85	0.80	0.80	0.90	0.90
M_2	2.20	1.00	0.80	1.60	0.90	0.90
M_3		1.00	0.85		1.25	0.90
M_4		1.50	0.80		1.75	0.90
M_5			1.20			1.20
M_6			1.60			1.60
M_7			2.00			2.00
M_8			2.40			2.40

In figure 3.1 is obtained from eq. (4),(5)and (6) considering initial step size as 1.From the plot we observe that there is a linear relationship between the previous step size and next step size .

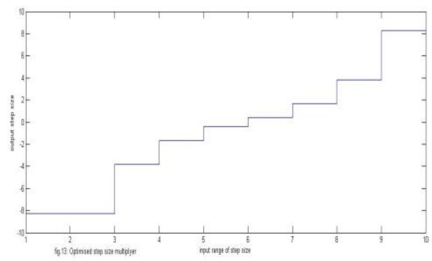


Fig 3.1 Relationship between input step size and adaptive step size

From graph of step size multiplier it is a critical that step size increase should be more rapid than step size decrease. In order to avoid any kind of granular or overload error this condition should be fulfilled necessarily.

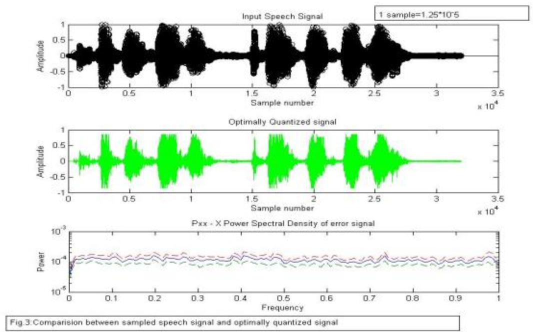


Fig3 Analysis of Adaptive pulse code modulated wave

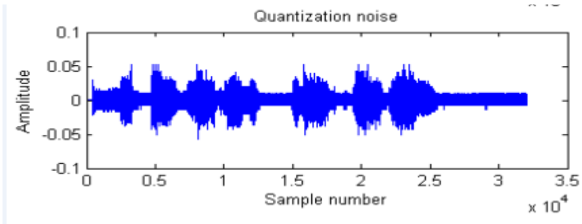


Fig3.1 Quantization noise analysis

If we take the difference between the input and quantized output at particular instant we will get a plot that is given by Fig 3.1. It is seen that max achieved signal error is around 0.05 that is 100 times lesser than maximum input signal applied.

Along with this from figure 3 power spectral density of quantized signal is fixed for all values of frequency at a value of 10^{-4} watts and the corresponding quantized output is appropriate for highly correlated signals as one that we have used for our work .

So the main conclusion that can be drawn from figure 4 is that the variation ofSNR with respect to quantization bits is exponential and becomes constant at35 db/decade if number of bits are increased after 10 bits . One more thing that can be observed from this plot is that in previous cases the value ofSNR for cases that we considered before were having value of SNR as a negative value whereas in this scheme we wave observed that it is positive since the starting which makes designing very much flexible as per the design of quantizer is concerned and correspondingly we can improve the signaling rate at a greater extent.

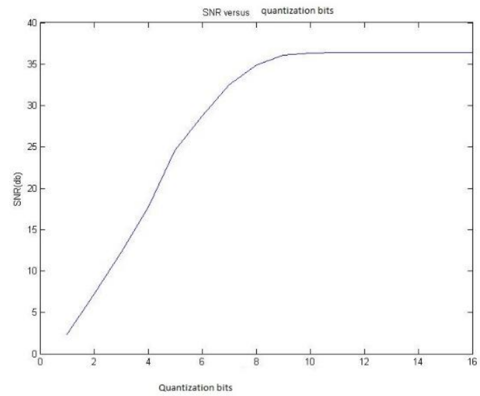


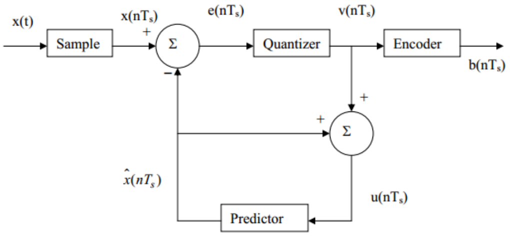
Fig 4 : Relationship between SNR and quantization bits for APCM scheme

4.Differential Pulse code modulation

Differential pulse code modulation (DPCM) as the name suggests is a procedure of transforming an analog into digital domain by taking the difference of the samples rather than considering whole sample.DPCM code words represent differences between samples unlike PCM where code words represented a sample value. As we know that the non-stationary speech signal has a high amount of correlation with that past signal so taking the difference we will deal with the redundancy that has occurred hence encoding those redundancy will reduce the bandwidth required to a larger extent.

In order to realizebasic concept (described above) is based on a technique in which we have to predict current sample value based upon previous samples (or sample) and we have to encode the difference between actual value of sample and predicted value (the difference between samples can be interpreted as prediction error).Because it's necessary to predict sample value DPCM is form of predictive coding.

DPCM compression depends on the prediction technique, well-conducted prediction techniques lead to good compression rates, in other cases DPCM could mean expansion comparing to regular PCM encoding.



Block Diagram1 for DPCM scheme

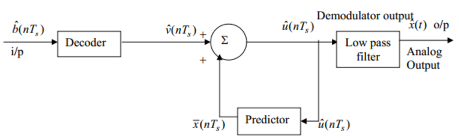
$\hat{x}$ (nTs) isprediction of the n-th input sample x(nTs).

ep(nTs) is n-th input to quantizer i.e.x(nTs) - $\hat{x}$ (nTs).
eq(nTs) is quantizer output for n-th prediction error.

ep(nTs)=eq(nTs)+q(nTs). (7)

u(nTs)= $\hat{x}$ (nTs)+eq(nTs)=x(nTs)+e(nTs) (8)

Equation 8 shows that u(nTs) is indeed a quantized version of x(nTs) . For a good prediction ep(nTs) will usually be small compared to x(nTs) and q(nTs) in turn will be very small compared to. Hence, the predictor unit should be so designed thatvariance of q(nTs) < variance of ep(nTs) << variance of x(nTs)[5] .



Block diagram2 for DPCM demodulator

u(nTs) is indeed a quantized version of x(nTs)therefore u(nTs) is demodulated output of x(nTs) with a quantisation error of q(nTs).

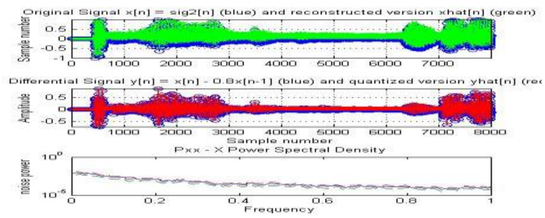


Fig 5: Analysis of DPCM scheme

Now in fig.5 we have compared the input speech signal with dpcm wave.

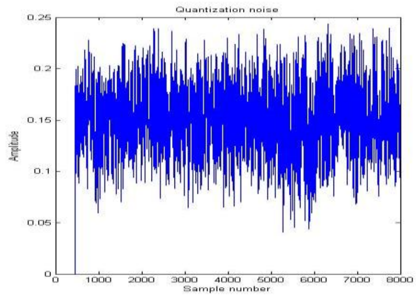


Fig 5.1 Quantization noise in DPCM signal.

In first subplot blue colored signal is our input speech signal and green one is demodulated signal. So we can see it traces the input signal quite well. According to our block diagram given above xhat(n)(reconstructed signal at output) is u(n). Insecond subplot we have plotted the error i.e. y(n)[eq(n) in above diagram] in bluecolour and quantised y(n)=yhat(n)[eq(n) in above diagram] in red colour.

Both the plots we have plotted against the sample number(number of samples=32000).

As we know that there is no substantial relation between number of bits required for quantization and SNR in case of DPCM[7]but wefoundthat there is a direct relation between the prediction coefficient andSNR as can be observed from figure 6.

This leads us to the fact that if we are going to track the predicted output we need the autocorrelation value produced by the prediction filter was closer to 1.This is true since as much correlated predicted signal is much lesser will be the difference hence lesser the number of bits required and hence less amount of quantization error.

But from the error spectrum that we got in figure 5 it can be seen that along with the reduction in number of bits the noisealso increases as compared to APCM wave which is a figure of demerit(maximum error in APCM was 0.05 here it is 0.2) .This must be because of some drastic change in amplitude which leads to a high value of difference that is provided to the quantizer.Since we are using a uniform quantizer it leads to the approximation error .Therefore we can conclude that we need to have a much robust technique that can deal with this problem. As we have seen in section 3 that adaption can be an option to eliminate such problem so in next section we will deal with adaptive approach towards DPCM.

Also there are no fluctuations in power of noise at lesser frequencies which makes it more preferable over APCM.

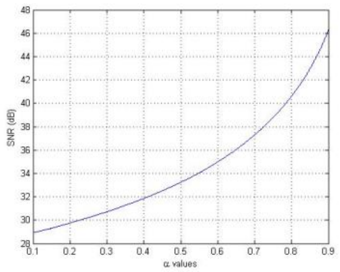


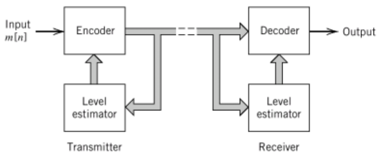
Fig 6 Relationship between SNR and autocorrelation coefficient of prediction filter

5. Adaptive Differential Pulse code modulation

As we have discussed in last section that in order to increase the value of SNR for the same number of bits we need to go for a scheme which adjusts the value of the value of quantizer based on the previous values of step sizes. This is what we will be doing in Adaptive Pulse Code Modulation.

Need for coding speech at low bit rates, we have two aims in mind:

1. Remove redundancies from the speech signal as far as possible.
2. Assign the available bits in a perceptually efficient manner.



Block diagram3 Adaptive quantization with backward estimation (AQB).

Adaptive quantizer

The adaptive quantizer that we are going to deal with has the same motivation as that of the one that we have used in APCM .The variance of prediction error which is the quantization input in DPCM is proportional to the input signal variance and input signal variance is either known or highly non stationary .The variance can be calculated if the signal is locally stationary .Strohdescribes straightforward estimation process for Gaussian signal[9]

However the one that we used for APCM explicitly use underload and overload cues to form adaptation logic.

Adaptive predictors

The adaptive filter is designed using Levinson Durbin algorithm [10] according to which linear prediction analysis is performed once per speech frame using the autocorrelation method with a 30 ms asymmetric window. The autocorrelation coefficients of windowed speech are computed and converted to LP coefficients using the Levinson algorithm every 80 samples (10ms). The LP coefficients are then transformed to line spectrum pairs for quantization and interpolation. The interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters for each subframe. This algorithm can be implemented by using a direct inbuilt function present in MATLAB i.e. “dpcmopt” this function provides us with an optimized value of quantizer as well as predictor parameters which can be directly implemented.

The only difference which is there in ADPCM from DPCM is the use of optimized parameters.

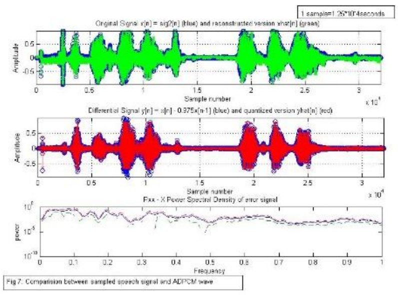


Fig 7: Comparison between sampled speech signal and ADPCM wave

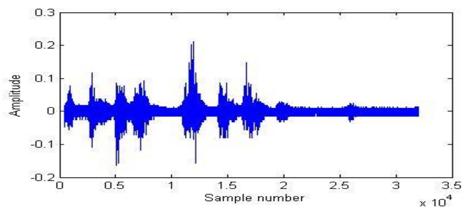


Fig 7.1:Quantization noise of ADPCM wave.

From figure 7 it is observed that due to adaptation in quantizing parameters there is a huge amount of correlation between the differential and quantized signal. Also the power spectral density becomes constant for high frequency signals but for low frequency signal it is very high. From figure 7.1 we found the average value of error to be $2.019*10^{-4}$V which is far more less than DPCM . Also the optimized prediction parameters were found to be 1 and 0.95 for given input speech signal which isquite good as per the auto correlation function is concerned. Since here the value of quantizer level is changing so the number of bits can also vary and hence the value of SNR.

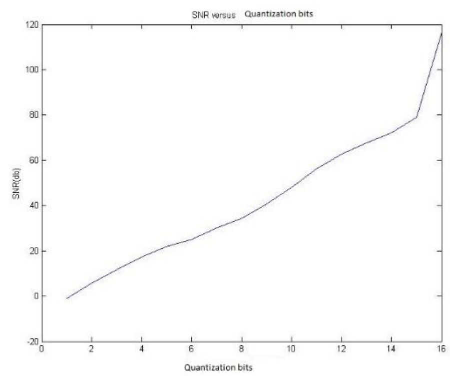


Fig 8 Relationship between SNR and Quantization bits for ADPCM scheme

If we talk about SNR(figure 8) till number of bits are 15 there is rise in SNR which is linear in nature but afterwards it changes steeply. Therefore ADPCM is the best technique as perquality is concerned

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6. Linear Delta Modulation

Delta Modulation is a techniquewhich provides a staircase approximation to an over-sampled version of the messagesignal (analog input). Samplingis at a rate higher than the Nyquist rate to increase the correlation between adjacent samples that simplifies quantizing of the encoded signal.

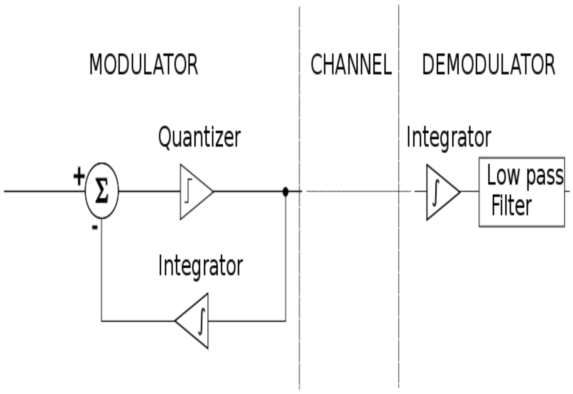
Principle Operation

- Difference between the input and the approximation is quantized in two levels - +/-Δ
- These levels correspond to positive/negative differences

Provided signal does not change very rapidly the approximation remains within +/-Δ

Derived forms of delta modulation are continuously variable slope delta modulation, delta-sigma modulation, and differential modulation. Differential pulse-code modulation is the super set of DM.

Rather than quantizing the absolute value of the input analog waveform, delta modulation quantizes the difference between the current and the previous step, as shown in the block diagram .



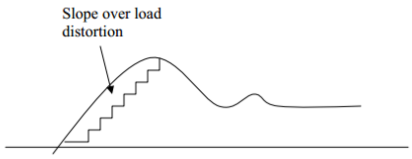
Block diagram4 for Delta modulator circuit

The modulator is made by a quantizer which converts the difference between the input signal and the average of the previous steps. In its simplest form, the quantizer can be realized with a comparator referenced to 0 (two levels quantizer), whose output is *1* or *0* if the input signal is positive or negative. It is a 1 bit-quantizer as it quantizes only a bit at a time. The demodulator is simply an integrator (like the one in the feedback loop) whose output rises or falls with each 1 or 0 received. The integrator itself constitutes a low-pass filter

SOURCES OF NOISE:

The two sources of noise in delta modulation are "slope overload", when steps are too small to track the original waveform, and "granularity", when steps are too large.

Slope overload distortion: if the input amplitude changes fast, the step by-step accumulation process may not catch up with the rate of change.



Slope overload distortion

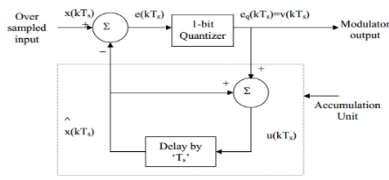


Figure9.1Elaborated Block diagram for delta modulation

Some interesting features of Delta Modulation • No effective prediction unit – the prediction unit of a DPCM coderis :

eliminated and replaced by a single-unit delay element.

- A 1-bit quantizer with two levels is used. The quantizer output simply indicateswhether the present input sample x(kTs) is more or less compared to itsaccumulated approximation $\hat{x}$(kTs) .
- Output $\hat{x}$(kTs) of the delay unit changes in small steps.
- The accumulator unit goes on adding the quantizer outputwith the previousaccumulated version $\hat{x}$(kTs) .
- u(kTs), is an approximate version of x(kTs).
- Performance of the Delta Modulation scheme is dependent on the sampling rate.

Most of the above comments are acceptable only when two consecutive input
samples are very close to each other.

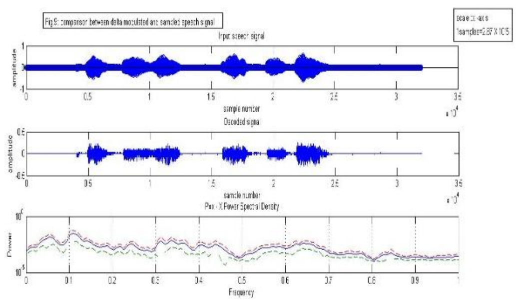


Fig 9.2: Analysis of delta modulated signal

This optimum (SNR-maximizing) step size dcan be related to the rms value of the firstdifferences in the input signal through a rule-of-thumb formula due to Abate [12]

d(opt)=abs(rms(m(g+1)-m(g))*log(2*F)) (9)

F=fs/(2*W) (10)

W=2500(for voice signals)

Fs(sampling frequency) varies from 100 to 1000.

where m(g) is the value corresponding to first non zero value of our voice signal.
So byvaringthe sampling frequency we will see how the max SNR will vary.

From fig.10 we can see the relation between snr and sampling frequency

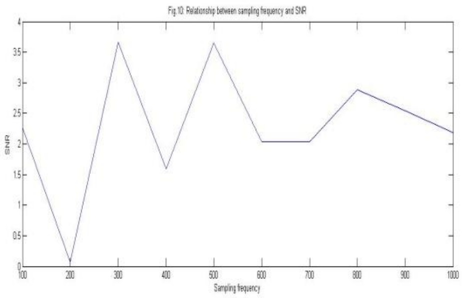


Figure 10: Comparison between SNR and sample frequency

From that we can conclude that max snr for this particular audio signal is at fs=300 and at fs=500.

d=0.0166 for fs=300 and d=0.0126 for fs=500 are the best delta to get maximum snr for this particular signal.These values will vary from signal to signal as it is 1 bit quantizer so its depends on the signal wheather it contains rapid changes or not.

We did it for sinusoidal signal also and SNR in that was 25db because it don't involve rapid changes so not much occurance of granular noise and slope overload.

We can see the input speech signal and decoded signal and power spectrum density of error in fig.9. We have take audio signal of 4 sec so number of samples =32000, so we have plotted input and decoded speech signal vs sample number.

This decoded signal is for fs=1000 i.e. d=0.0072 that give the snr=2.

So we can see that it nicely traces the input signal but there is some error in the magnitude part that can be reduced by choosing right delta and corresponding sampling frequency as delta modulation depends on only two things delta size and sampling frequency.

But we can also say that nature of signal matters a lot in case of delta modulation.

7. Adaptive delta Modulation

Another type of DM is Adaptive Delta Modulation (ADM). In which the step-size isn't fixed. The step-size becomes progressively larger when slope overload occurs. The gain of the amplifier is adjusted in response to a control voltage from the SAMPLER, which signals the onset of slope overload. The step size is proportional to the amplifier gain. This was observed in an earlier experiment. Slope overload is indicated by a succession of output pulses of the same sign.

The SAMPLER monitors the delta modulated signal, and signals when there is no change of polarity over 3 or more successive samples. The actual ADAPTIVE CONTROL signal is +2 volt under 'normal' conditions, and rises to +4 volt when slope overload is detected.

The gain of the amplifier, and hence the step size, is made proportional to this Control voltage. Provided the slope overload was only moderate the approximation will 'catch up' with the wave being sampled. The gain will then return to normal when output samples shows different polarities.

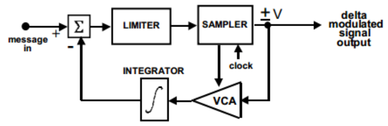


Fig 11.1 block diagram for adaptive delta modulation

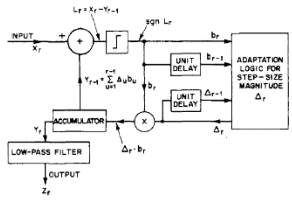


Fig 11.2 : Block diagram showing the ajdustments in delta that we have used

SNR is only affected by the initial step size as after that it will adapt by adjusting the step size.

Ina typical, and conceptually least complex, realization, described by Jayant [15], successive bits *b,* and *b,-l* are compared to detect probable slope overload (e(n)=e(n-1)) or probable granularity (e(n)!=e(n-1)).Slope overload and granularity tend to correspond, respectively, to occurrences of like and unlike (successive) bits, The specific adaptation rule in [15] is the following:

d(n)=d(n-1)*P.^(e(n)*e(n-1))

where d is the step size and e(n) is subtraction of current and previous value

The rate of step-size increase (or decrease) is given by a single factor P. Notice that P= 1 represents (nonadaptive) LDM.
For adaptive :

1<= P <=2

Here we have taken P as 2.

So we will vary initial size from 0 to 1 with the intervals of 0.1 see how the SNR will change or wich initial step size is best suited for that particular speech signal.As the signal varies thw SNR will also vary as the deciding factor is how well or fastly it can adjust its step size according to the input signal given.

This change in SNR due to diferent initial step size can be seen in fig.12.

Here the SNR is maximum at d=0.3.SNR value is not that high becoz of our selection of initial step size if we take interval as 0.01 so we will see increase in SNR .

We have done it for sinusoidal also and in that case we got maximum SNR at 0.1 i.e 25 db.

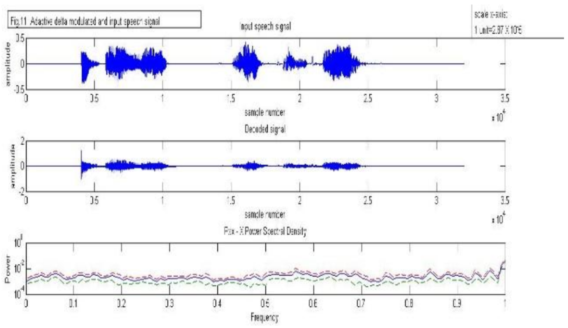


Figure 11.3 Analysis of ADM wave

From fig.11.3 we can see the input speech signal, decoded signal, power spectrum density of error signal. We have take audio signal of 4 sec so number os samples =32000 , so we have plotted input and decoded speech signal vs sample number.

This decoded signal is for initial delta =1 that give the snr=0.6.

This is less because from 1 it had to adjust itself to the given signal so it causes lot of error initially.

Inspite of giving that much high initial step size you can see in decoded signal that how well it traces the signal afterwards.

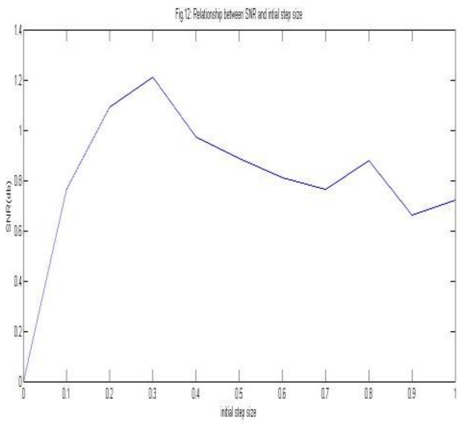


Fig 12 Relationship between SNR and initial step size of ADM scheme

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