Performance Analysis of DPCM and ADPCM

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ABSTRACT

The advantages of using signals in digital domain are of many folds. Some of the advantages compared to the analog signals are multiplexing, storage, compression & ease of reproduction of digital signals. Added to this the Moore's law factor, the cost of digital hardware continues to halve every two years while performance or capacity doubles over the same period has led to an exponential use of devices that are digital in nature.

Digital signals are obtained by sampling & quantizing the analog signal so that they can be efficiently represented. In this paper, different kinds of waveform coding techniques such as DPCM & ADPCM are studied. Performance is evaluated based on Signal to Quantization Noise Ratio (SQNR) & Mean square error (MSE) measures. Encoding & decoding complexity as a function of time is also studied.

General Terms

Speech Processing, Waveform Coding Techniques, Statistical Signal Processing

Keywords

DPCM, ADPCM, SQNR, MSE, Predictor, Adaptive Quantizer

1. INTRODUCTION

The communication model requires the source to be represented efficiently so that the available bandwidth is effectively utilized. This requires processing of signals in digital domain which are obtained by sampling & quantization of an analog signal.

Speech being an integral part of communication needs to be represented effectively. One of the earliest techniques used to represent a digitized speech is Pulse code Modulation (PCM). PCM is a natural extension of representing an analog signal. Even though PCM found wide spread use in telephone industry, it used a large number of bits to represent a signal sample & hence required a lot of bandwidth. Hence alternate methods were developed so as to utilize bandwidth effectively.

The speech samples are highly correlated & using this property several schemes have been developed. In one of the scheme, instead of sending the quantized sample,difference between the successive samples is quantized & transmitted. The samples at the receiver can be got by adding the received sample with the previous decoded sample. Since the range of difference samples are small compared to the original samples, they can be encoded using a fewer number of bits thereby reducing the transmission bandwidth required [1].

Further improvement can be made by predicting the sample values using past sample values & thereby reducing the range of the difference signal. This scheme is known as DPCM in which the predictor is adaptive. By making the quantizer adaptive, dynamic range of the quantizer can be modified which results in ADPCM.

The layout of the paper is as follows. Section 2 describes the DPCM & ADPCM in brief. Section 3 gives the performance analysis of DPCM & ADPCM separately. Section 4 presents the conclusions.

2. DPCM & ADPCM

DPCM & ADPCM as discussed in [1], [2]&[3] are based on prediction of the sample value using past sample values. The general encoder & decoder structure for predictor based scheme is shown in figure 1.

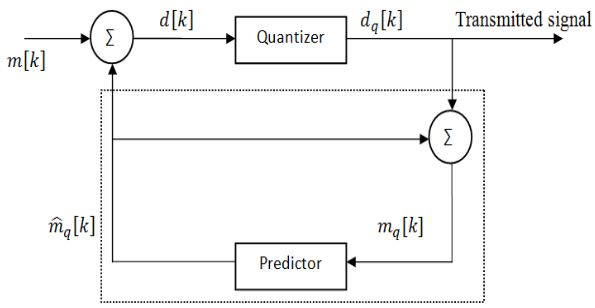

Figure 1: Encoder & Decoder for predictor based schemes

The analog speech signal $m(t)$is sampled at Nyquist rate to obtain speech samples $m[k]$. Predictor predicts the current sample value using past sample values to generate $\hat{m}_q[k]$. The difference between the sample & its predicted value is quantized & transmitted. The relations for encoder are summarized below

$$d[k] = m[k] - \hat{m}_q[k] \quad (1)$$

$$d_q[k] = d[k] + q[k] \quad (2)$$

The lower part of the encoder structure forms the decoder whose structure is outlined in figure 1. The received signal $d_q[k]$ is passed through the predictor to produce estimate $\hat{m}_q[k]$ of speech samples. The decoded signal $m_q[k]$ is given by

$$m_q[k] = d_q[k] + \hat{m}_q[k] \quad (3)$$

The decoded signal has always quantization noise $q[k]$ associated with it.

2.1 Differential Pulse Code Modulation (DPCM)

In DPCM, predictor is adaptive in nature. The N^{th} order predictor predicts the sample value $\hat{m}[k]$ using past N values of $m[k]$. The linear predictor as in is one of those in which

$$\hat{m}[k] = a_1 m[k-1] + a_2\, m[k-2] + \cdots a_N m[k-N] \quad (4)$$

$$i.e\ \hat{m}[k] = \sum_{i=1}^{N} a_i\, m[k-i] \quad 1 \le i \le N \qquad (5)$$

The predictor coefficient a_i can be optimized using two methods as in [4]:

i)Forward Adaptive Prediction: The samples are grouped into blocks & autocorrelation coefficient for each block is calculated. The predictor coefficients are obtained from autocorrelation coefficients. This requires buffering of input which introduces delay & additional bandwidth is required for transmitting side information.

ii)**Backward Adaptive Prediction**: The parameters are adapted based on output of the encoder. Since encoded data is available to decoder, transmission of side information is not required.

If the input samples are assumed to be stationary, using backward adaptation strategy predictor coefficients a_i can be optimized for better prediction with minimum mean square error criterion (MMSE).

The predictor coefficients subjected to MMSE criterion can be described using matrix notation as $\mathbf{A=R^{-1}*R0}$

$*$denotes the matrix multiplication

A is column matrix of order (N x 1) consisting of a_j

R is square matrix of order (N x N) consisting of elements R_{ij}, R_{ij} being the autocorrelation of the i^{th} sample with the j^{th} sample

R0 is a column matrix of order (N x1) of elements R_{0i}, R_{0i} being the autocorrelation with the i^{th} sample.

2.2 Adaptive Differential Pulse Code Modulation (ADPCM)

In DPCM unbounded inputs can cause overload noise. Unbounded inputs occur since the overall dynamic range of the difference samples are unknown, which largely depends on how closely the predictor predicts. Since overload noise is more serious than the granular noise which occurs within the bounded input, the range of the quantizer should be properly fixed. This is done through the adaptive quantizer & the scheme is called as ADPCM.

In adaptive quantizer parameters of the quantizer are dynamically adjusted based on the two schemes [4]:

i)Forward Adaptive Quantization: Source output is divided into blocks of data, which is analyzed before quantization & parameters are set accordingly. The settings of quantizer are transmitted as side information.

There are few downsides associated with Forward Adaptive Quantization:

1. Coding delay is involved in processing the block.
2. Compression ratio is reduced due to transmission of side information.
3. Small block sizes capture changes in the input statistics but require frequent transmission of side information whereas large block size reduces side information transmission but does not capture variations in signal statistics.

ii)**Backward Adaptive Quantization**: Past quantized samples are used to adapt quantizer parameters. The step sizes are adapted based on where the previous quantized samples fall.

Backward Adaptive quantization requires past quantized samples to be monitored. One of the simple schemes is to consider the most recent quantized output. This quantizer is commonly known as Jayant Quantizer [5], [6].

Jayant Quantizer assigns multipliers for each interval & based on the most recent quantized output, step sizes & hence reconstruction levels are modified according to corresponding multipliers.

If we assign multiplier M_k for the k^{th} interval then the step sizes are adapted according to the equation $\Delta_{n+1}=M_n\Delta_n$; where Δ is the step size &'n' is the index for the last quantized sample.

The step size multipliers are greater than unity for outer levels & less than one for inner levels. Step size multipliers [7] are also symmetric in nature as shown in table 1 & it is required to specify maximum & minimum step sizes so that step size remains within operational limits.

Table 1: Multiplication factors for step size adjustment for different bits per sample

	2	3	4
M(1)	0.80	0.90	0.90
M(2)	1.60	0.90	0.90
M(3)	-----	1.25	0.90
M(4)	-----	1.70	0.90
M(5)	-----	-----	1.20
M(6)	-----	-----	1.60
M(7)	-----	-----	2.00
M(8)	-----	-----	2.40

3. RESULTS

The main objective of study was to show performance improvement provided by DPCM & ADPCM as compared to PCM. The performance metrics considered were:

1. Variation of Signal to Quantization Noise Ratio (SQNR) as function of bit rate.
2. Mean Square error (MSE) between the output (decoded) & input samples, input & predicted samples.
3. Average encoding & decoding time as a function of predictor order (M).

The performance metrics were analyzed by considering an arbitrary duration of speech samples of bandwidth 4 KHz & sampled at Nyquist frequency of 8 KHz. Analysis were done from 8-64 Kbps for DPCM & 16-32 Kbps for ADPCM as a function of predictor order.

The analysis of speech samples was also done using pre added street noise & wideband noise for which similar improvements in results were obtained. Properties of test speech samples are specified in table 2. Figure 2 & 3 show the speech samples & its spectrum respectively.

Table 2: **Properties of Speech Samples**

Parameter	Value
File format	Wave
Bit rate mode	Constant
Bit depth	16 bits
Bit rate	128 kbps
Channels	1

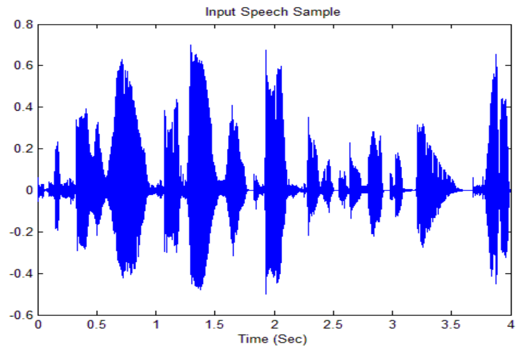

Figure 2: Speech sample of duration 4 seconds

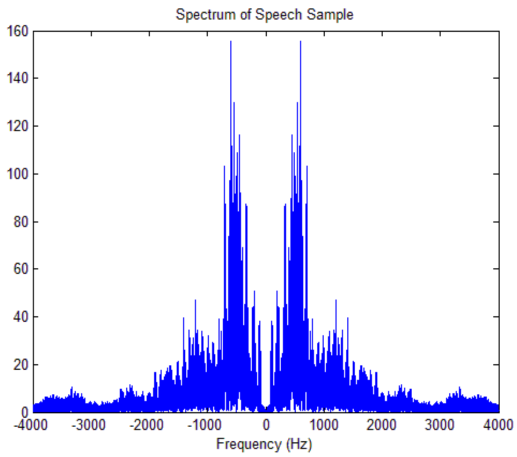

Figure 3: Spectrum of Speech samples

3.1 Analysis of DPCM

DPCM was designed using a linear predictor in backward prediction mode subjected to MMSE criterion. Quantizer used is a uniform quantizer with an assumption that the dynamic range of the difference signal considered is 50% of the dynamic range of the input samples.

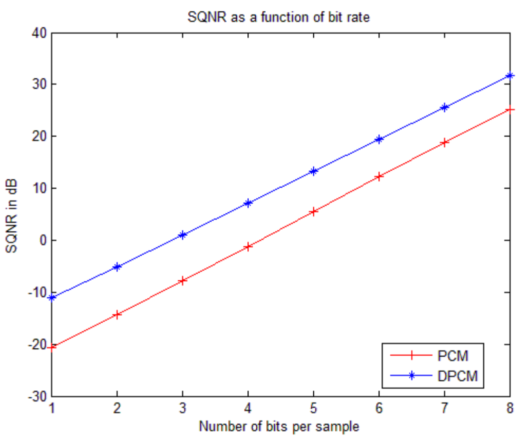

Figure 4: SQNR as function of bit rate for DPCM

SQNR improvement in DPCM is largely due to removal of redundancy present in PCM as shown in figure 4. DPCM can provide 5-8 dB improvement in SQNR for a fixed bit rate or for a given SQNR can result in lower bit rate.

SQNR variation with predictor order as function of bit rate remains marginal as shown in table 3. This depends largely on the statistics of the source under consideration.

Table 3: SQNR (dB) variation with predictor order (row) & number of bits per sample (column)

	1	2	3	4	5	6	7	8
2	-11.149	-5.110	0.968	7.066	13.262	19.474	25.576	31.662
3	-11.264	-5.171	0.935	7.036	13.225	19.444	25.549	31.647
4	-11.139	-4.995	1.124	7.224	13.390	19.508	25.538	31.220
5	-11.241	-4.974	1.155	7.277	13.395	19.513	25.665	31.707
6	-11.285	-4.890	1.196	7.331	13.408	19.540	25.649	31.725
7	-11.338	-4.878	1.244	7.362	13.477	19.547	25.655	31.687
8	-11.701	-4.783	1.282	7.338	13.436	19.597	25.672	31.717

MSE provides a better test for predictor order as shown in figure 5 & 6. MSE in both cases converges at higher predictor order due to less prediction error. It is also evident from the plots that MSE between the output & input samples is more than the MSE of difference signal due to the quantization noise present in the decoded signal.

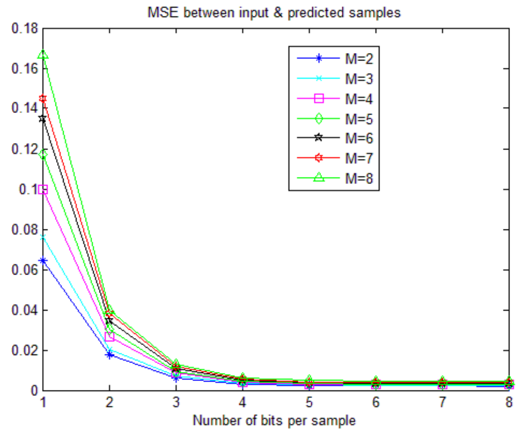

Figure 5: MSE of difference signal in DPCM

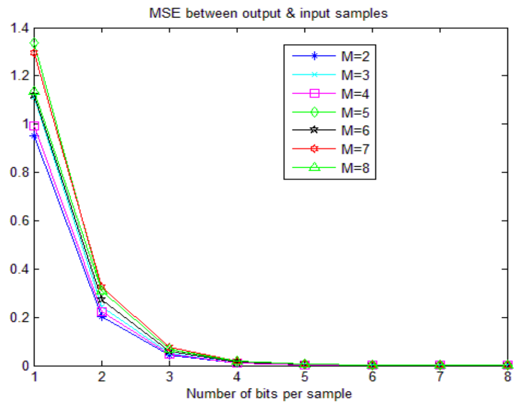

Figure 6: MSE of decoded samples in DPCM

The average encoding & decoding time required are shown in figure 7 & 8 respectively. Encoding & decoding time are considered for relative measures only & hence do not present actual processing times.

Encoding & decoding time increase linearly with prediction order but remain constant across the quantizer levels. This is due to uniform quantizer which results in reconstruction levels of the quantizer being fixed but varies across the predictor order as more processing of samples are required to give predictor output. Since only predictor is used in decoding, decoding times are relatively less compared to encoding time.

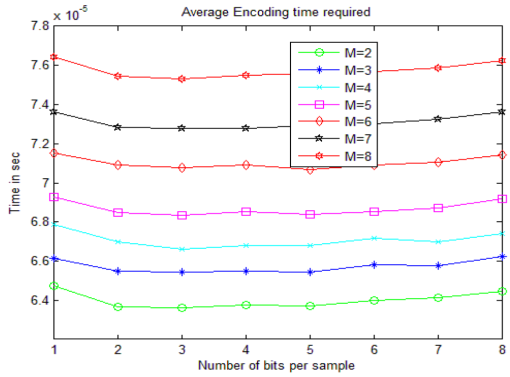

Figure 7: Average encoding time for DPCM

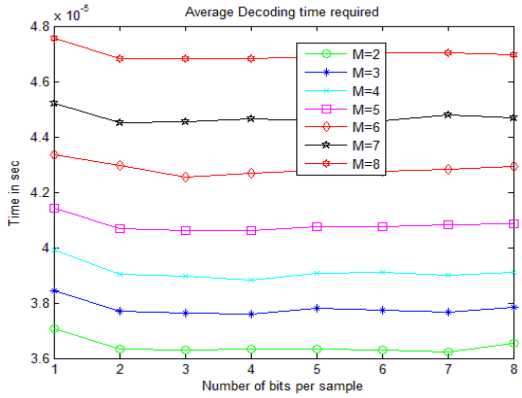

Figure 8: Average Decoding time for DPCM

The decoded speech samples using seventh order predictor & 4 bits per sample is shown in figure 9. Low pass filtering the speech samples with cut off frequency equal to one half of Nyquist frequency gives the reconstructed speech signal.

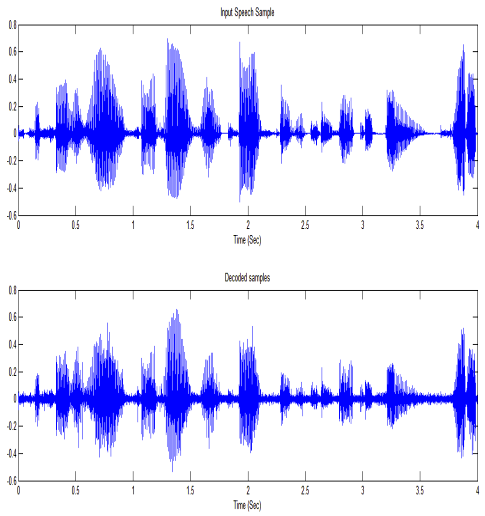

Figure 9: Speech & its reconstructed samples using DPCM

3.2 Analysis of ADPCM

ADPCM design uses the backward adaptive prediction as in DPCM with an adaptive quantizer. Jayant quantizer with multipliers as specified in table 1 are used. Maximum & minimum step sizes were fixed suitable for operation.

The SQNR plots for ADPCM as shown in figure 10 show linear increase in SQNR as bits per sample is increased. The increase in SQNR is due to the variation of dynamic range of the quantizer which results in better handling of overload noise. For a given SQNR, ADPCM coded samples can be represented using 4 bits as compared to 8 bits per sample as in PCM. Thus bandwidth savings of 50% or more is easily achievable in ADPCM.

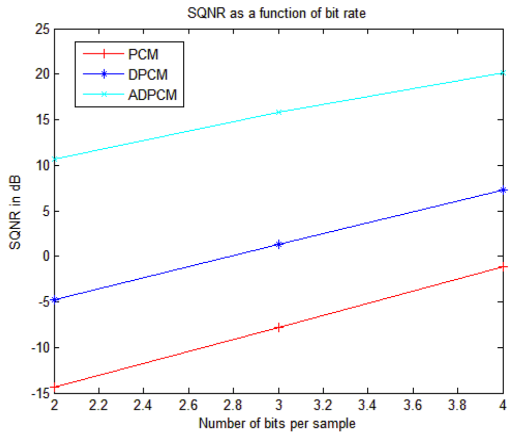

Figure 10: SQNR as function of bit rate for ADPCM

The SQNR increase with predictor order is marginal as with DPCM & it appears constant across predictor orders.

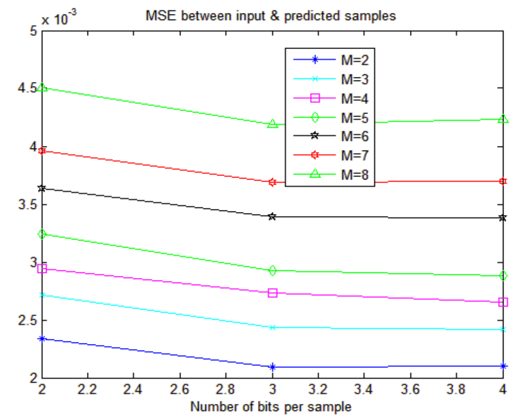

Figure 11: MSE as a function of predictor order & bit rate

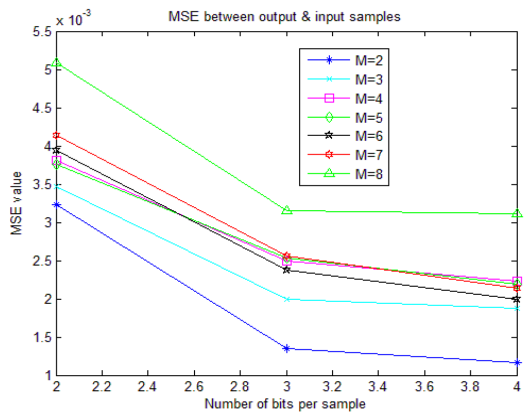

Figure 12: MSE as a function of predictor order & bit rate

The mean square error plots for ADPCM are shown in figure 11 & 12. MSE of the difference signal is small compared to MSE between the input & output samples. The MSE values increase with predictor order but can converge at higher bit rates & it is usually better to go with higher predictor orders since they produce close estimate of the samples.

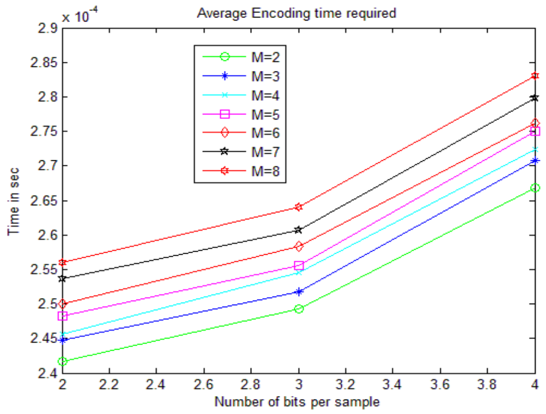

Figure 13: Average encoding time for ADPCM

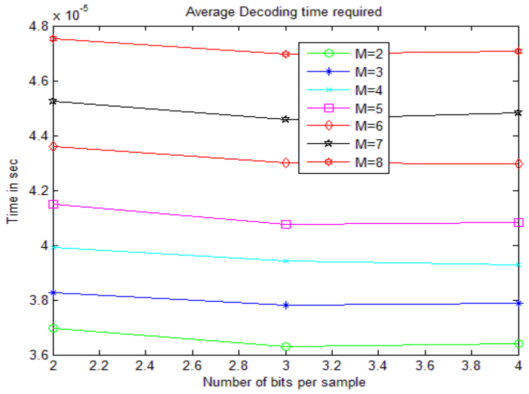

Figure 14: Average decoding time for ADPCM

The encoding & decoding times of ADPCM are shown in figures 13 & 14 respectively. The encoding times increase linearly with both predictor order & bits per sample. As adaptive quantizer adapts during every sample based on the previous output, mapping of the next sample values will be different from previous time. This incurs as additional factor for increase in time. As in the case of DPCM, higher predictor orders take more time due to increase in processing of samples required to produce estimated output.

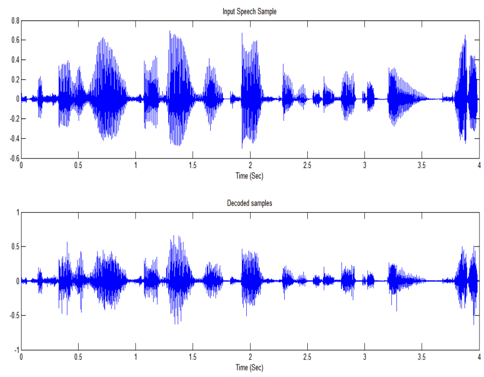

Figure 15: Speech & its reconstructed samples using ADPCM

The decoding times remain constant across bits per sample as the decoding structure does not make use of the adaptive quantizer thereby simplifying the decoding process.

Decoded speech samples using sixth order predictor & 2 bits per sample are shown in figure 15.The samples are amplified & low pass filtered with cut off frequency equal to one half of the Nyquist frequency to obtain the analog speech waveform.

4. CONCLUSIONS

The analysis of DPCM & ADPCM show huge improvement in SQNR & reduction in bandwidth required as compared to PCM. ADPCM, with its better dynamic range handling offers significant improvement in performance as compared to DPCM.

The general inferences that can be drawn from the above results are:

1. Backward adaptive predictors & quantizer are usually preferred in design as forward adaptive algorithms introduce delay &require more bandwidth which is not acceptable in connections having multiple links.
2. It is better to use backward adaptive predictors of higher order so as to obtain a better estimate of the samples which also results in low MSE values.
3. Adaptive quantizer used in ADPCM should be chosen such that the reconstruction levels used does not lose speech intelligibility. This is necessary as adaptive quantizer contributes significantly to the encoding times required.

5. ACKNOWLEDGMENTS

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