The BRITISH UNIVERSITY IN EGYPT

DIGITAL COMMUNICATIONS (1) PROJECT

ADAPTIVE PCM AUDIO TRANSMISSION SYSTEM

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1. Introduction

The standard approach of digitally representing sampled analog signals is Pulse Connection Modulation (PCM). It is the basis of digital telephony and audio. However, transmission channels rarely stay consistent in the real world, they are of variable Bandwidth (BW) and Signal-to-Noise Ratios (SNR). In order to preserve the integrity of communication in real world, an Adaptive PCM system is necessary.

This project will be based on designing and modelling with MATLAB, an adaptive baseband PCM system. The system will take an audio file and send it on an Additive White Gaussian Noise (AWGN) channel and reconstruct it to the receiver. The main characteristics of this design are:

- Bandwidth Adaptation: The system is dynamically adjusted to adjust the bit resolution (ℓ) according to the channel bandwidth.
- SNR Adaptation: The receiver uses an adaptive detection threshold in order to counter the channel noise.
- Expanding "SQNR Improvement": A-law compandor is used to enhance the Signal-to-Quantization-Noise Ratio (SQNR), especially when dealing with low amplitude signals.

Project Objectives:

Rather than only sending data, this project aims to:

- Model the complete Transmitter and receiver chain: Compressor $\rightarrow Quantizer \rightarrow Encoder \rightarrow Channel \rightarrow Pulse\ Detection \rightarrow Decoder \rightarrow Expander$
- Introduce an algorithm, which computes the most optimal bit depth (l) in respect of input bandwidth.
- Compare the performance of the system in various channel conditions (High Quality vs. Low Quality) based on SQNR, Bandwidth Efficiency and Normalized Mean Squared Error (NMSE).

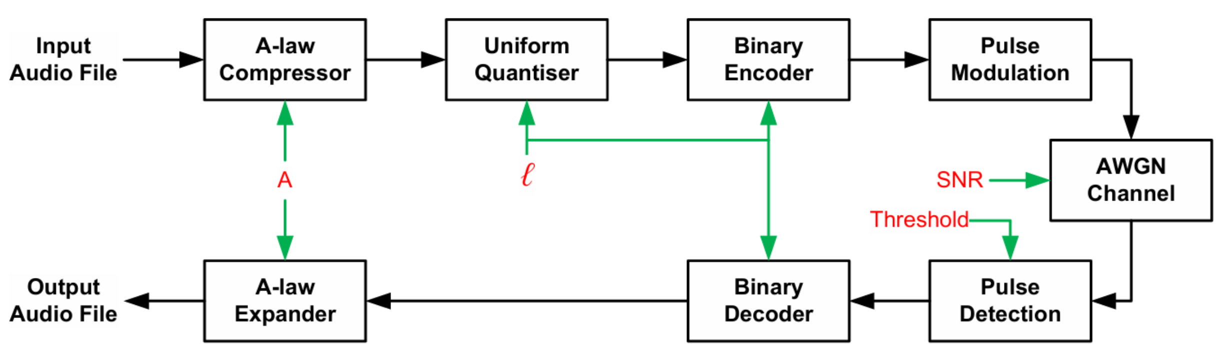

Figure 1 Block diagram of the required system

2. Required Tasks

Write a MATLAB code that models the system presented above. The program asks the user to enter the following data:

- Audio file name.
- Available transmission BW in KHz.
- Channel SNR in dB.
- The value of the “A” parameter used for A-law companding.

Accordingly, the program reads the audio file, extracts the sample rate, selects the suitable parameters (bit resolution and detection threshold), transmits the data through the AWGN channel and receives them at the receiver side. After it is done, the program should display the following:

- Selected sample bit resolution, and resulting transmission bit-rate.
- Bandwidth efficiency.
- SQNR.
- NMSE between the transmitted and received audio clips. For a transmitted n-length sequence x and received sequence $\hat{x}$, NMSE is defined as: $NMSE = \frac{\sum_{i=1}^{n}(x_i - \hat{x}_i)^2}{\sum_{i=1}^{n}(x_i)^2}$
- Play the received audio file.

3. System Design & Code Segment Analysis

In this part, a developed MATLAB model is analyzed and divided into functional parts which are correlated with the theoretical blocks.

3.1. Logic of Parameter Adaptation

The main characteristic of this system is the possibility to adjust the number of bits per sample (l)- dependent on the bandwidth available. MATLAB Implementation:

```
% adaptation

BW_Hz = BW_kHz * 1000;
l = floor((2 * BW_Hz) / fs);
```

Theoretical Analysis:

The Nyquist criterion of the zero Inter-Symbol Interference (ISI) is the theoretical transmission limit of a baseband channel of bandwidth BW, where that $R_b \leq 2 \cdot BW$. As $R_b = \ell \cdot f_s$ then the maximum number of bits is calculated as $\ell = \left\lfloor \frac{2 \cdot BW}{f_s} \right\rfloor$. This is to make sure that the signal fits within the channel.

3.2. The Transmitter (Compression) The Transmitter (Quantization)

The A-law Companding before uniform quantization is applied to enhance the SQNR of speech signals (which have large values of small values). MATLAB Implementation:

```
% A-law Compressor
K = 1 + log(A_val);
x_comp = sign(x) .* log(1 + A_val * abs(x)) / K;

% Uniform Quantizer
partition = linspace(-1, 1, L+1);
codebook = linspace(-1, 1, L);
partition = partition(2:end-1);

[index, x_quant] = quantiz(x_comp, partition, codebook);
```

Theoretical Analysis: The code applies the A-law compression law:

$$y = \frac{1 + \ln(A|x)}{1 + \ln(A)} sgn(x) \ \ for \ \frac{1}{A} \leq |x| \leq 1$$

These compressed continuous values are then quantized into $L = 2^l$ discrete values by the quantiz function. Lastly, dec2bin is used to translate the indices in decimal to a binary stream to be sent.

3.3. The AWGN Channel

The channel is assumed to be an Additive White Gaussian Noise channel. MATLAB Implementation:

```
%% AWGN Channel

rx_signal_noisy = awgn(signal_tx, SNR_dB, 'measured');
```

This operation introduces Gaussian noise to the signal in order to produce the desired SNR_dB. This is a simulated interference of nature.

3.4. The Receiver (Expansion and Adaptive Threshold)

The noisy signal has to be recovered by the receiver. MATLAB Implementation:

```
% Pulse Detection

adaptive_threshold = mean(rx_signal_noisy);
rx_bits_detected = rx_signal_noisy > adaptive_threshold;
```

Theoretical Analysis:

The optimum decision threshold is a bipolar signaling scheme (where $+V \rightarrow 1, -V \rightarrow 0$) the mid-point. The system adapts the threshold to any offset in the SC or signal drift, based on the calculation of the mean(rx_signal_noisy) and the Bit Error Rate (BER) is minimized. Once the bits have been recovered and converted to quantized values, the bits are recompressed with the inverse logarithmic formula by the A-law Expander.

4. Simulation Scenarios & Results

We modeled two different scenarios Scenario A (High Quality) and Scenario B (Low Quality) to test the system. The findings show the adaptability of the system to adjusting the concerned ($\boldsymbol{\ell}$) and the ensuing effect on the performance measures.

Table 1 Scenarios Comparison Table

Metric	Scenario A: High Quality	Scenario B: Low Quality
Input Bandwidth	270 kHz	70 kHz
Input SNR	40 dB	15 dB
Bit Resolution ($\boldsymbol{\ell}$)	12 bits	3 bits
Transmission Rate R_b	529.20 kbps	132.30 kbps
Bandwidth Efficiency	1.96 bps/Hz	1.89 bps/Hz
BW Utilization	98.00 %	94.50 %
SQNR	65.58 dB	9.73 dB
NMSE	0.000001	1.099130
Audio Quality	100.00 %	19.47 %

5. Visual Analysis & Interpretation

To discern the performance of the system, we monitored the plots of both scenarios of simulation:

5.1. Scenario A: High Quality Case

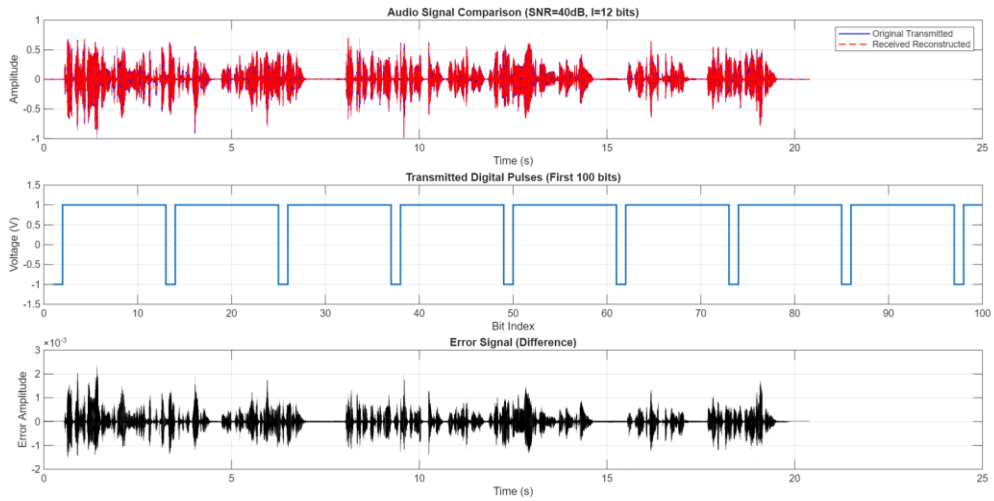

Figure 2 Scenario A: High Quality Case outputs

- System Response: Because the bandwidth that was available was huge (270 kHz), the system was able to compute a very high bit resolution of $\ell = 12$ bits.
- Visual Observation: As can be seen in the Audio Signals plot, the received signal (red dashed line) and the transmitted signal (blue line) coincide at every point. The error signal is virtually zero.
- Interpretation: This ideal reconstruction is made possible by the fact that 12 bits give 4096 quantization levels, and this means that quantization noise is removed. The output is a high-fidelity audio of 100% Quality Score.

5.2. Scenario B: Low Quality Case

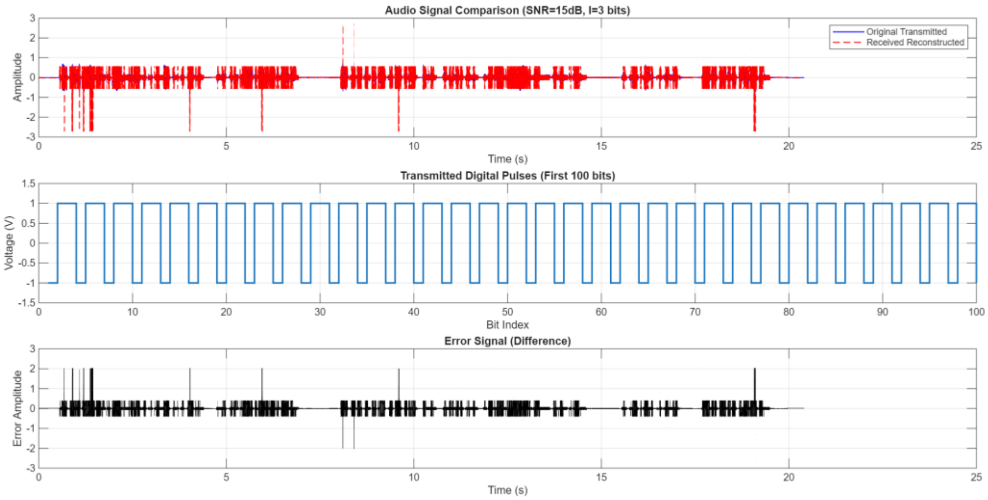

Figure 3 Scenario B: Low Quality Case outputs

- System Response: The bandwidth had been limited to 70 kHz only and this compelled the system to scale down the resolution to $\ell = 3$ bits to accommodate the data.
- Visual Observation: The distortion of the signal can be easily seen in the plots. The waveform received is not smooth, but in the form of a staircase.
- Interpretation: This distortion is made possible by the fact that only 8 levels of voltages can be represented in 3 bits, which is not enough to display the audio details.
- Critical Insight: Even though the system was very efficient (94.5% channel utilization), the audio quality decreased to 19.47. This determines that high efficiency does not imply good quality in case the capacity of the channel itself is insufficient.

6. Conclusion

In this project, we have managed to develop and model an Adaptive PCM system in MATLAB. The outcome of the simulation revealed that there is a basic correlation between channel resources and signal quality. We noted that although the system has managed to adjust the bit depth ($\boldsymbol{\ell}$) to prevent Inter-Symbol Interference (ISI) in different bandwidths, a high bandwidth efficiency did not always imply a high audio quality. In Scenario B we used 94.5 percent of the channel capacity, but the tight bandwidth limitation caused the bit depth to be very low, which caused a lot of quantization noise. This proves that effective audio delivery needs a tradeoff between an effective code and the adequate physical channel resource.