Lab10 Report

李梓源　SID:12211225

李沅朔　SID:12210301

Introduction

This lab course mainly focus on μ-law quantization and adaptive quantization, which automatically fits the smaller signal amplitude. In this lab, we also learns how to encode a speech using μ-law from bottom to the end and do several comparision between μ-law quantiziation and static quantization, which shows several advantages of encoding the speech siganal adaptively.

Problem 1

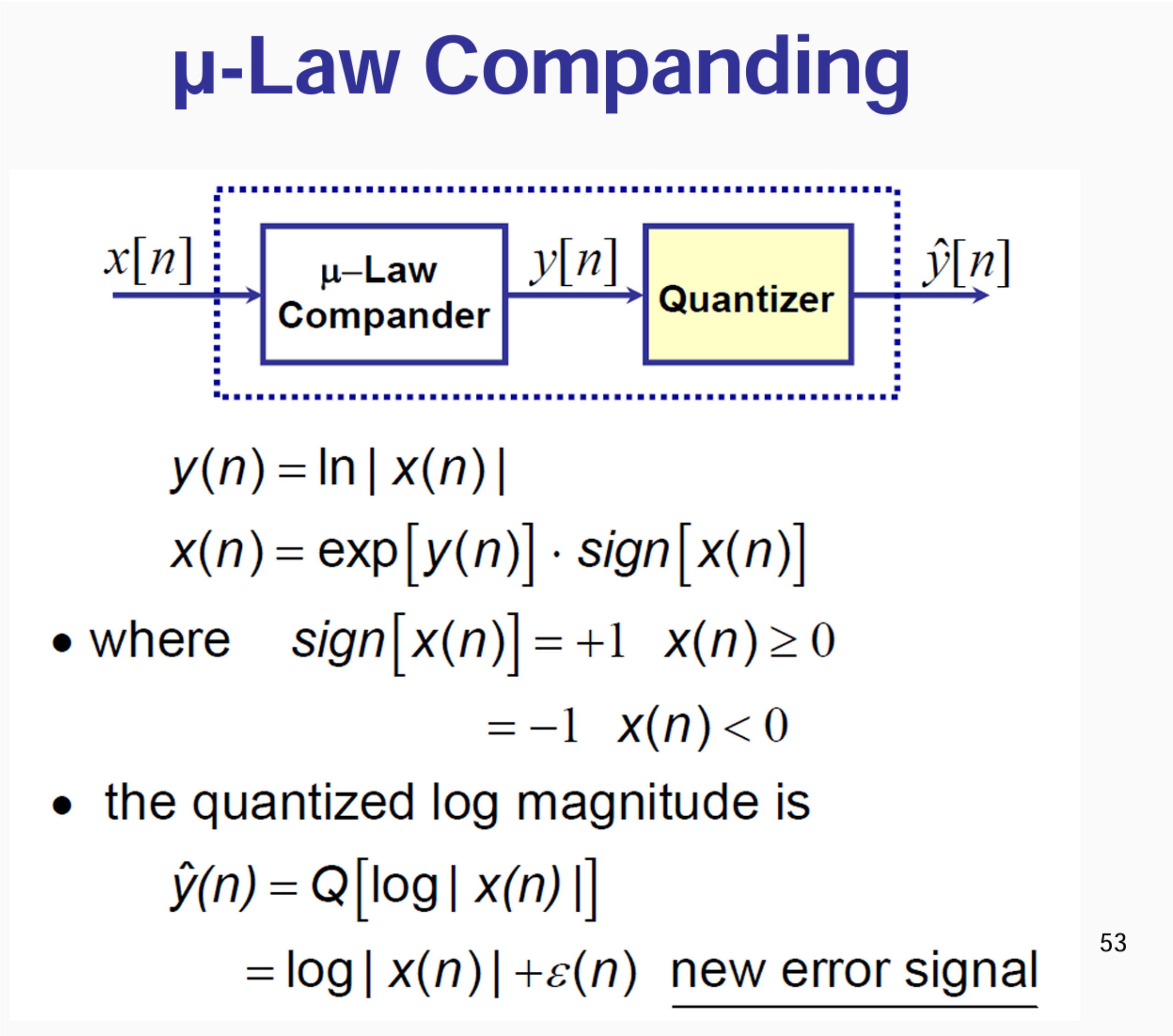
Problem description

1. Plot the output waveform y[n] of the μ - law compressor and plot a histogram of the output samples.
2. Write an m - file for the inverse of the μ - law compressor, named mulawinv(y, mu). The function should follow a specific calling sequence and parameter definition. Test the inverse system by applying it to the output of mulaw() without quantization.
3. Compute and plot the first 8000 samples of the resulting quantization error, plot a histogram of the quantization error amplitudes, and plot the power spectrums of the resulting quantization errors.

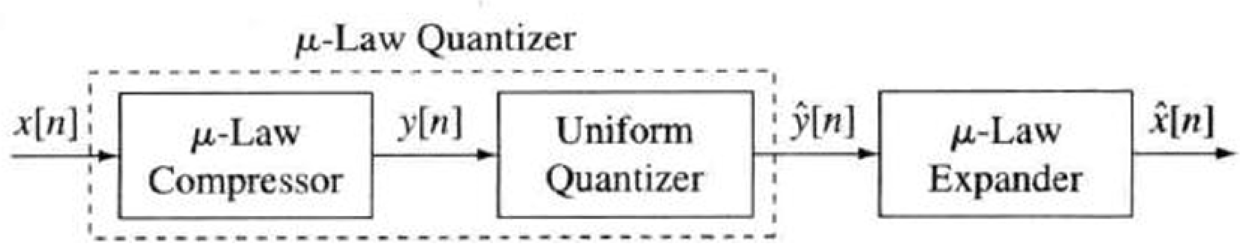
Solutions and process

1. Use the given MATLAB function `mulaw()` to do the μ compression

2. Follow the steps shown in the figure, first calculate the amplitude, and then recover the sign by multplying `sign(y)`



3. First calculate yhat using MATLAB function `fxquant()`, then calculate the error spctrum using `pspect()`



Key code segment

1.

```
x = (-1:0.001:1);
figure
plot(mulaw(x,1), 'LineWidth',1), hold on;
plot(mulaw(x,20), 'LineWidth',1), hold on;
plot(mulaw(x,50), 'LineWidth',1), hold on;
plot(mulaw(x,100), 'LineWidth',1), hold on;
plot(mulaw(x,225), 'LineWidth',1), hold on;
plot(mulaw(x,500), 'LineWidth',1), hold off;
title('Waveform After Encoding')
legend('1', '20', '50', '100', '225', '500');
```

2. Derive function `my_mulawinv()`

```
[y, fs] = audioread('s5.wav');
y = y(1300:18800);
figure
subplot(211)
plot(y), title('Original Wavform')
subplot(212)
plot(mulaw(y, 225)), title('Encoded Waveform')
saveas(gcf, "D:/作业提交/大三 下/语音信号处理/lab10/P1_b_1.png", 'png')
% 直方图
figure
histogram(mulaw(y, 225))
title('Histogram of Encoded Waveform')
saveas(gcf, "D:/作业提交/大三 下/语音信号处理/lab10/P1_b_2.png", 'png')

function x = my_mulawinv(y, mu)
    % 计算幅度部分
    abs_x = ( (1 + mu) .^ abs(y) - 1 ) / mu;
    % 恢复符号
    x = abs_x .* sign(y);
end
```

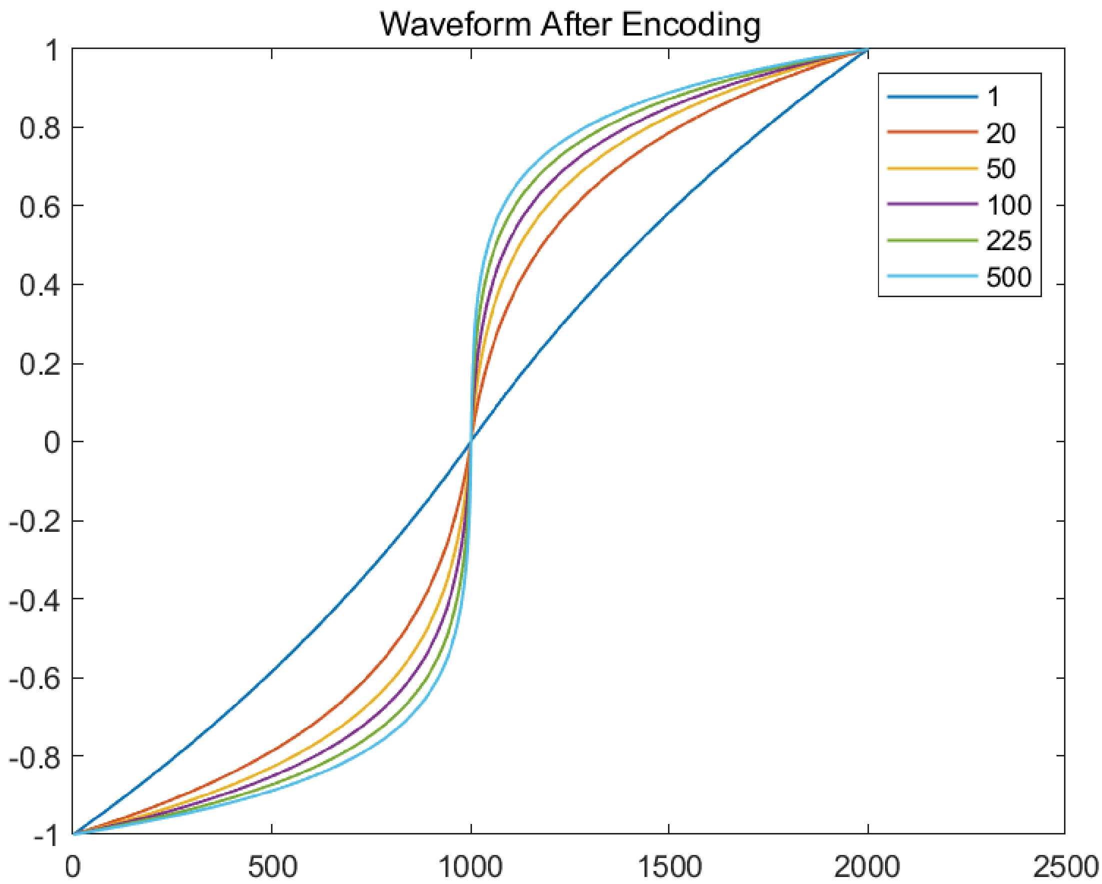
3. Plotting the error power spectrum

```
yh_4 = fxquant(mulaw(x, 225), 4, 'round', 'sat');
yh_8 = fxquant(mulaw(x, 225), 8, 'round', 'sat');
yh_10 = fxquant(mulaw(x, 225), 10, 'round', 'sat');
yh_6 = fxquant(mulaw(x, 225), 6, 'round', 'sat');
e_4 = mulawinv(yh_4, 255) - x;
e_6 = mulawinv(yh_6, 255) - x;
e_8 = mulawinv(yh_8, 255) - x;
e_10 = mulawinv(yh_10, 255) - x;
e_4 = e_4(1:8000);
e_6 = e_6(1:8000);
e_8 = e_8(1:8000);
e_10 = e_10(1:8000);
% histogram
figure
subplot(221), histogram(e_4), title('Quantization Error with 4 bits')
```

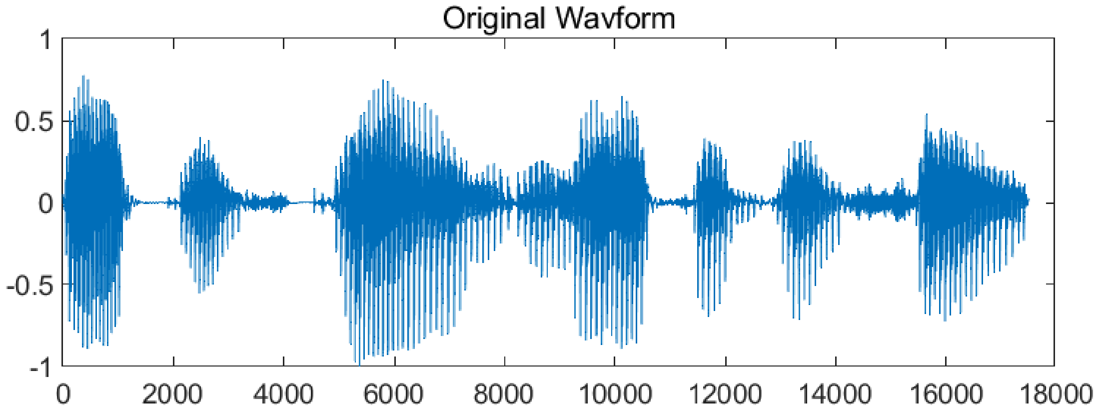
```
subplot(222), histogram(e_6), title('Quantization Error with 6 bits')
subplot(223), histogram(e_8), title('Quantization Error with 8 bits')
subplot(224), histogram(e_10), title('Quantization Error with 10 bits')
saveas(gcf, "D:/作业提交/大三 下/语音信号处理/lab10/P1_d_1.png", 'png')

% Power Spectrum
win_len = 512;
[p_4, f4] = pspect(e_4, fs, win_len, win_len);
[p_6, f6] = pspect(e_6, fs, win_len, win_len);
[p_8, f8] = pspect(e_8, fs, win_len, win_len);
[p_10, f10] = pspect(e_10, fs, win_len, win_len);
figure
plot(f4, p_4, 'LineWidth',1), hold on;
plot(f6, p_6, 'LineWidth',1), hold on;
plot(f8, p_8, 'LineWidth',1), hold on;
plot(f10, p_10, 'LineWidth',1), hold off;
xlabel('Frequency (HZ)');
ylabel('Power Spectrum (dB)');
legend('4-bit', '6-bit', '8-bit', '10-bit');
saveas(gcf, "D:/作业提交/大三 下/语音信号处理/lab10/P1_d_2.png", 'png')
```

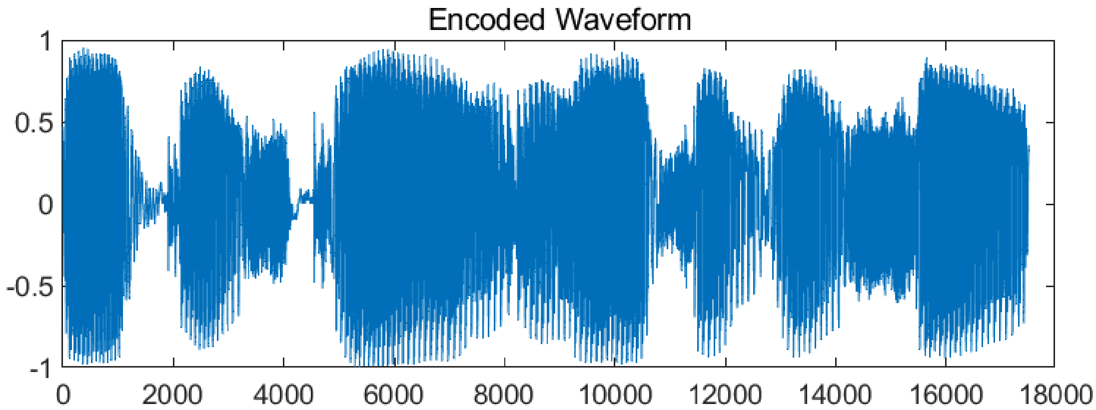
Result and Analysis

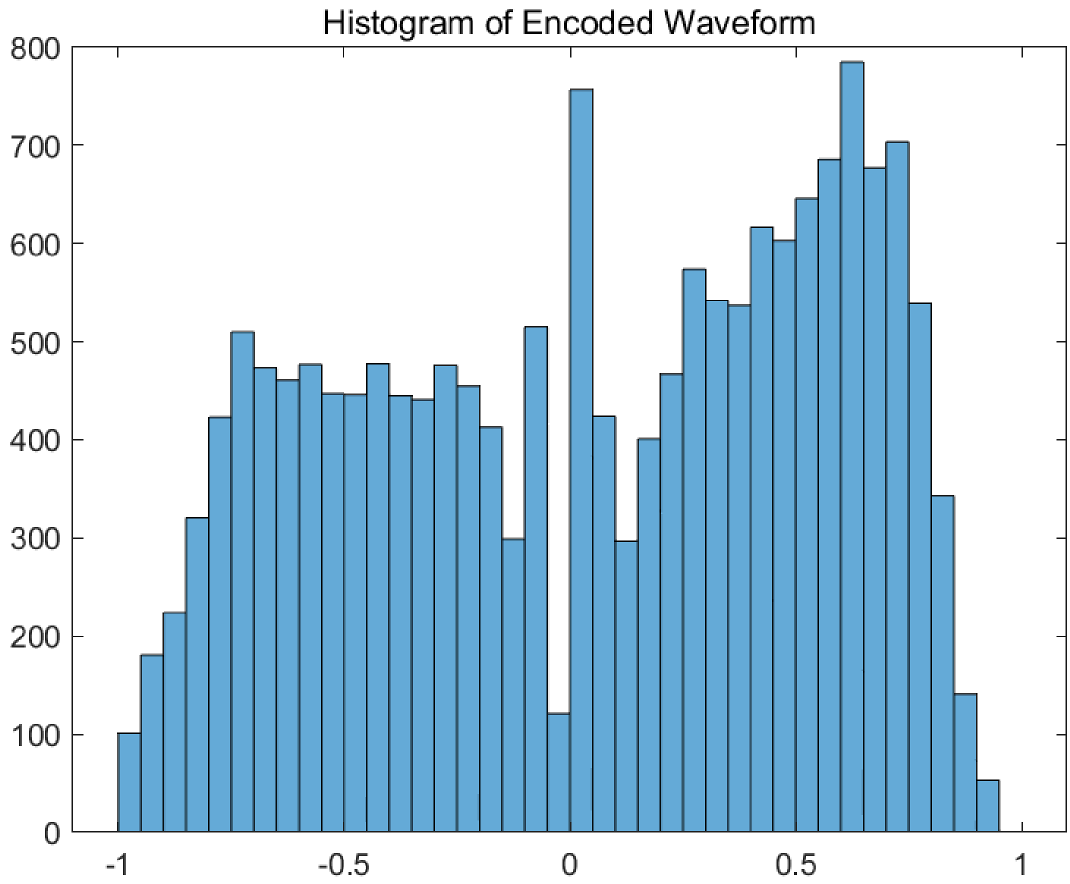
1. Question A



2. Question B







As we can see in the result, after encoding, the samll amplitude are increased, which provide convenience and accuracy for the following encoding and decoding.

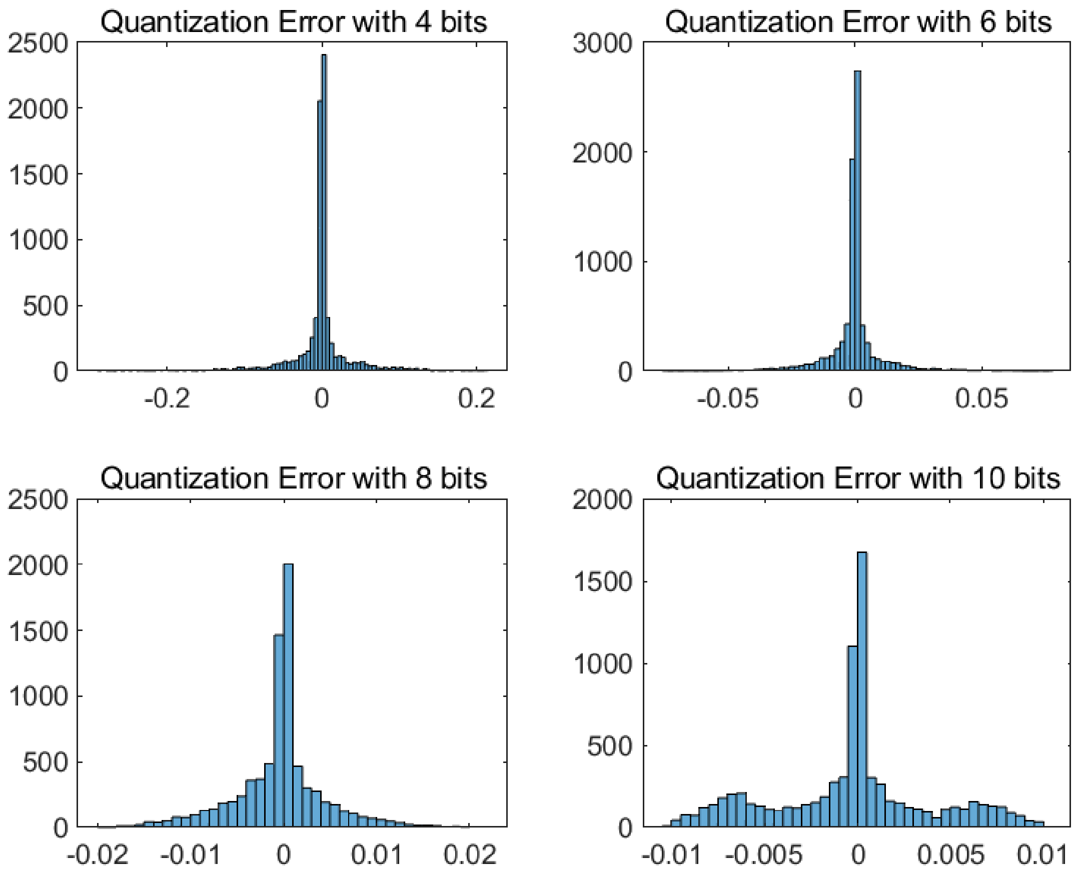
3. Question C

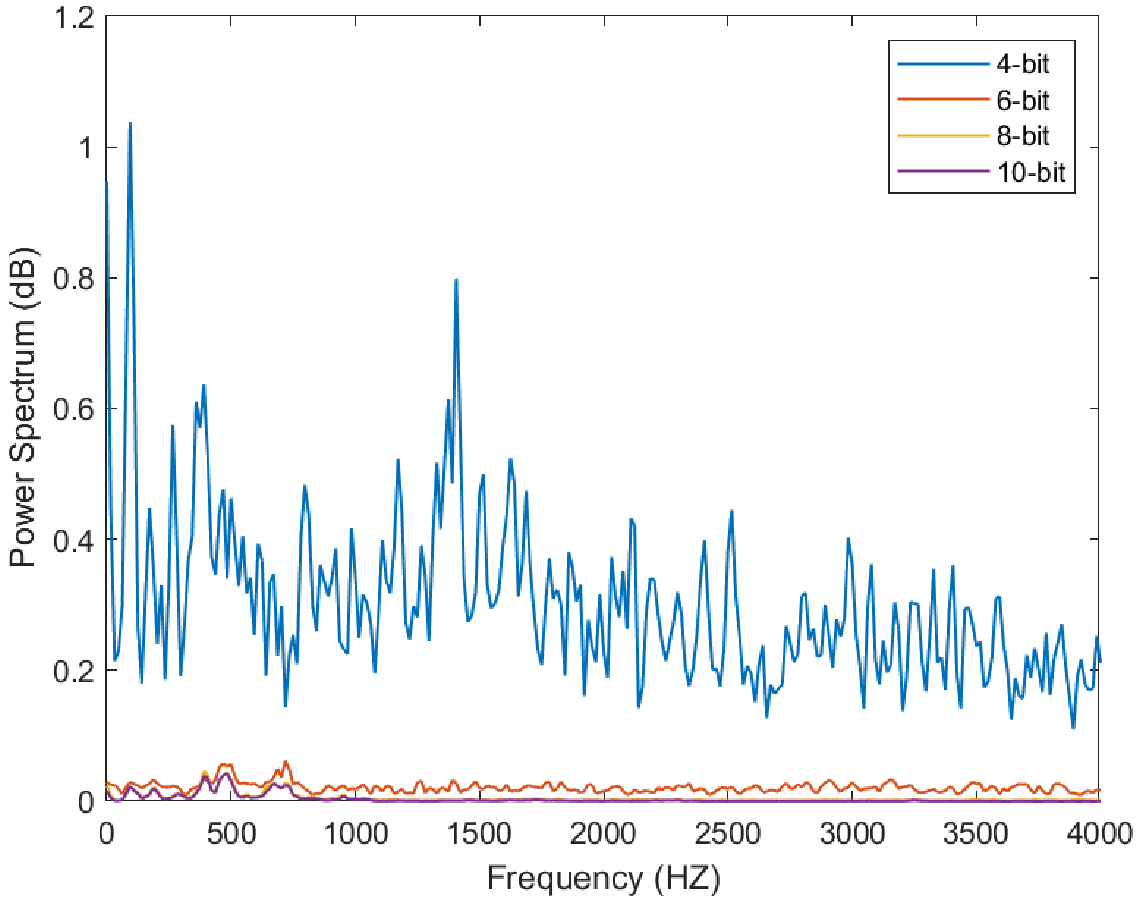
```
v = my_mulawinv(mulaw(y, 225), 225);
mean(v-y)
```

In the code, we use the above method to derive the gap between v and original siganl y. The result is 4.62*10^-19, which is samll enough to prove the effiency of `my_mulawinv()`

4. Question D

   We tested more situations aside form 6-bits(4, 8, 10 bits are all tested), and we can see that as the bit number increase, the error decrease evidently.





Problem 2

Problem description

1. Write function to compute SNR,output SNR in dB. Use it to compute SNRs for 8 - and 9 - bit uniform quantization of a s5.wav speech segment
2. Vary the input speech signal level and write a program to plot measured SNR for all 13 cases. Additionally, use differnet stastic quantization value and μ quantization value and compare its diffences.

Solution and process

1. Derive a MATLAB function `SNR` to calculate the SNR for a uniform B-bit quantizer. The mathmatical description is shown bellow:

$$\text{SNR} = 10\log\left(\frac{\sum_{n=0}^{L-1}(x[n])^2}{\sum_{n=0}^{L-1}(\hat{x}[n]-x[n])^2}\right).$$

2. Compare the quantization result for differnt parameters and both stastic quantization and μ-bit quantization. To be concise, in the μ-bit quantization, first compress the original signal, then do a uniform quantization, thirdly, expand the μ-bit encoded signal and finally calacualte the SNR.

Key code segment

1. Derive the SNR function based on the formula above

```
[y, fs] = audioread('s5.wav');
% mu-量化
y_new = mulaw(y, 225);
% 均匀量化
yh_8 = fxquant(mulaw(y, 225), 8, 'round', 'sat');
yh_9 = fxquant(mulaw(y, 225), 9, 'round', 'sat');
yh_10 = fxquant(mulaw(y, 225), 10, 'round', 'sat');
yh_11 = fxquant(mulaw(y, 225), 11, 'round', 'sat');
yh_12 = fxquant(mulaw(y, 225), 12, 'round', 'sat');
% s = y - yh_8
[snr_8, e_8] = SNR(yh_8, y_new);
[snr_9, e_9] = SNR(yh_9, y_new);
[snr_10, e_10] = SNR(yh_10, y_new);
[snr_11, e_11] = SNR(yh_11, y_new);
[snr_12, e_12] = SNR(yh_12, y_new);
bit = [8,9,10,11,12];
snr = [snr_8, snr_9, snr_10, snr_11, snr_12];
figure
stem(bit, snr), xlabel('Bits Number'), ylabel('SNR (dB)'), title('SNR for Uniform
Quatization')
grid on

function y = mulaw(x, mu)
    y = sign(x) .* log(1 + mu * abs(x)) / log(1 + mu);
end
```

```
function [snr, e] = SNR(xh, x)
    L = length(x);
    sum_sig = 0;
    sum_err = 0;
    for i=1:L
        sum_sig = sum_sig + x(i).^2;   % 分号！！！
        sum_err = sum_err + (x(i)-xh(i)).^2;   % 分号！！！
    end
    sum_sig
    sum_err
    snr = 10*log10(sum_sig/sum_err)
    e = mulaw(xh, 225) - x;
%     mean(e)
end
```

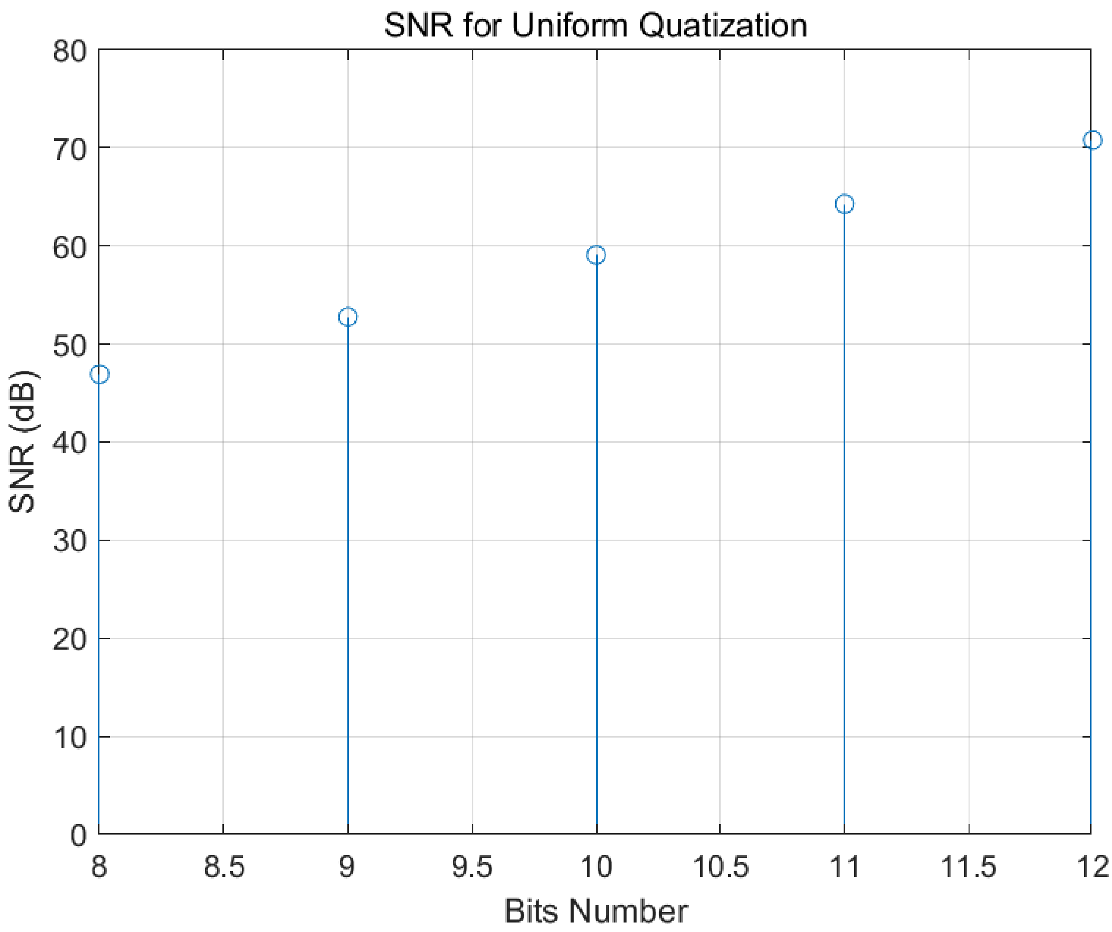
2. Compare all situations in one figure.

```
% 参数设置
factors = 2.^(0:-1:-12);
mu_values = [100, 255, 500];
bits_uniform = 6:10;

% 初始化存储矩阵
snr_uniform = zeros(length(factors), length(bits_uniform));
snr_mulaw = zeros(length(factors), length(mu_values));
inv_sigma_x = zeros(length(factors), 1);

% 计算不同幅度下的SNR
for i = 1:length(factors)
    x_scaled = x * factors(i);
    sigma_x = std(x_scaled);
    inv_sigma_x(i) = 1 / sigma_x;

    % 均匀量化
    for j = 1:length(bits_uniform)
        x_quant = fxquant(x_scaled, bits_uniform(j), 'round', 'sat');
        snr_uniform(i,j) = 10 * log10( sum(x_scaled.^2) / sum((x_scaled - x_quant).^2) );
    end

    % μ-law量化 (6 bits)
    for k = 1:length(mu_values)
        % μ-law压缩
        y = mulaw(x_scaled, mu_values(k));
        % 均匀量化（6 bits）
        y_quant = fxquant(y, 6, 'round', 'sat');
        % μ-law扩展
        x_recon = mulawinv(y_quant, mu_values(k));
        % 计算 SNR (dB)
        snr_mulaw(i,k) = 10 * log10( sum(x_scaled.^2) / sum((x_scaled - x_recon).^2) );
    end
end
```

```
figure;
semilogx(inv_sigma_x, snr_uniform, 'LineWidth', 1.5); hold on;
semilogx(inv_sigma_x, snr_mulaw, '--', 'LineWidth', 2);
xlabel('1/\sigma_x'); ylabel('SNR (dB)');
title('SNR vs 1/\sigma_x for Uniform and \mu-law Quantization');
grid on;

legend_labels = [...
    arrayfun(@(b) sprintf('%dbit', b), bits_uniform, 'UniformOutput', false), ...
    arrayfun(@(mu) sprintf('\\mu-law (\\mu=%d)', mu), mu_values, 'UniformOutput', false) ...
];
legend(legend_labels, 'Location', 'southwest');
```

Result and Analysis

1. As shown in the result, whenever one bit is added, the SNR increases approximatlly 6dB, which fit the tone.



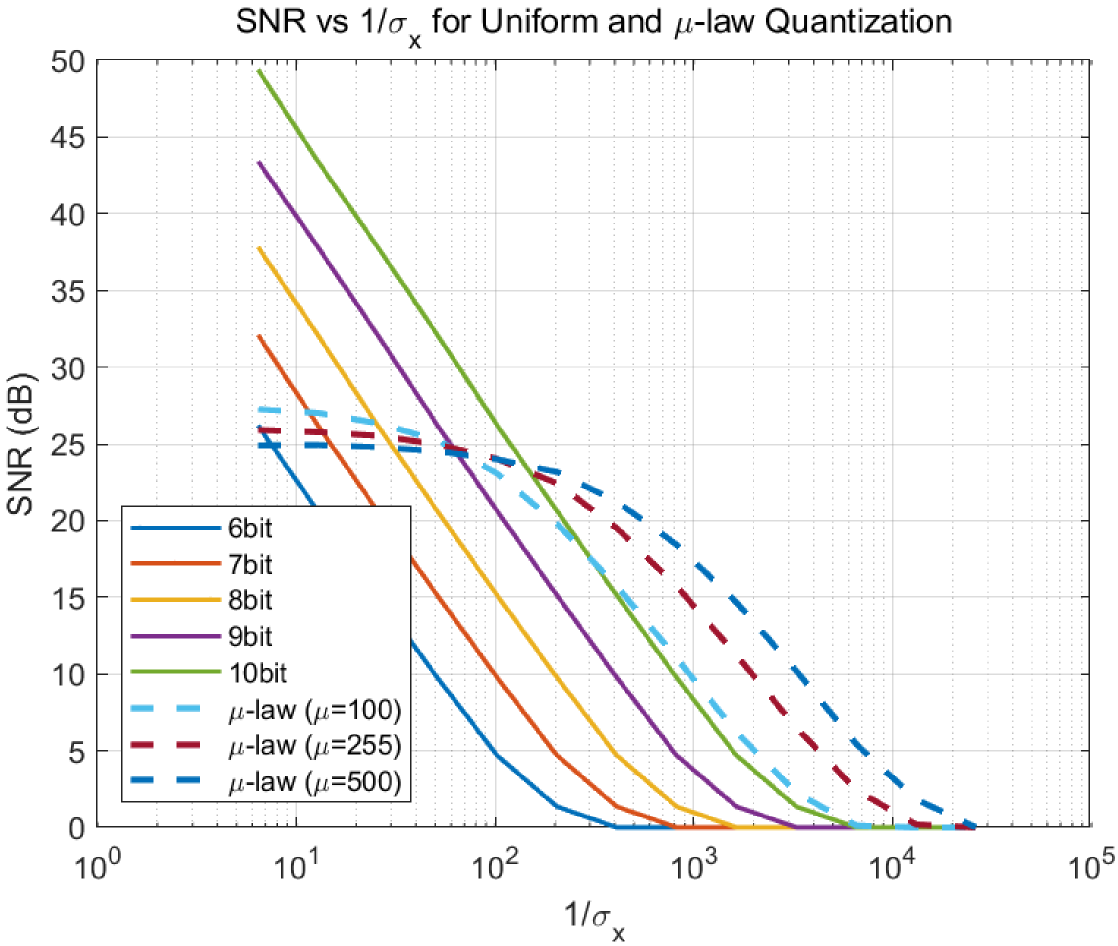
2. For μ-law quantization:

- When $1/\sigma_x$ varies the SNR remains relatively stable.
- Optimal performance is achieved at μ=255 (standard parameter for telephone speech coding).

3. For Uniform quantization:

- The SNR degrades sharply as the signal amplitude decreases (due to higher quantization errors for small signals).

- Approximately ~12 bits are required to match the dynamic range of 6-bit μ-law quantization.



Problem 3

- **Problem description:**
  This exercise focuses on demonstrating the process of adaptive quantization using both IIR and FIR filters. The task involves processing a given speech file, s5.wav, and calculating the standard deviation σ[n] of the signal.
- **Solution and Process:**

As the expressions of filters are already given, the coefficients for FIR and IIR are determined. We will use those coefficients and the `filter` function to calculate the deviation, and at last use those calculated deviation values to equalize signals.

Effects of different parameters (alpha and M) will be discussed.

- **Key code segment:**

1. The IIR filters are designed using b and a coefficients, and applied using `filter()`. Be careful that the input here is squared audio signal.

```
alpha_1 = 0.9;
alpha_2 = 0.99;

b_1 = [0 (1-alpha_1)];
b_2 = [0 (1-alpha_2)];
```

```
a_1 = [1 -alpha_1];
a_2 = [1 -alpha_2];

[aud, fs] = audioread('s5.wav');
aud = aud - mean(aud);
aud_squared = aud.^2;

delta_1 = sqrt(filter(b_1, a_1, aud_squared));
delta_2 = sqrt(filter(b_2, a_2, aud_squared));

gain_equalized_1 = aud ./ delta_1;
gain_equalized_2 = aud ./ delta_2;
```

Superimposed Plots are drawn using `hold on;` command.

```
figure(3)
plot(aud(2700:6700));
title('S5 Original Signal');
xlabel('Sample Number'); ylabel('Amplitude');
```

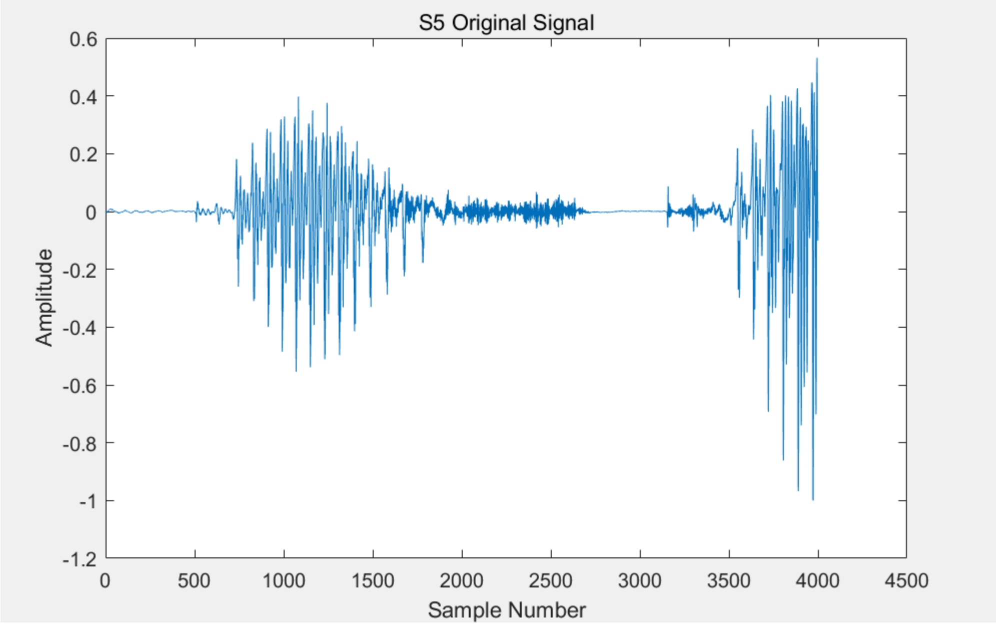
original waveform is drawn for comparison.

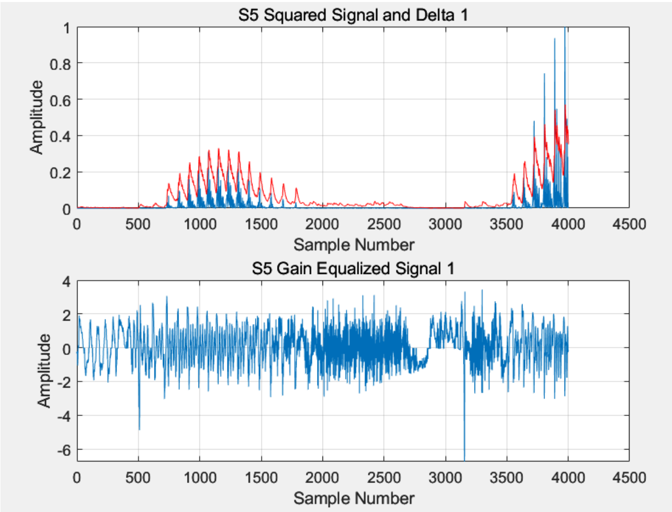
2. FIR filter is designed using a similar manner.

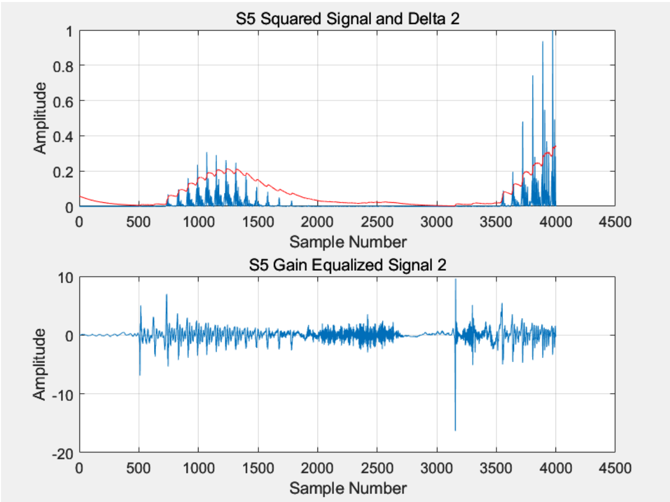
```
M_1 = 10;
M_2 = 100;

b_fir_1 = ones(1, M_1) / M_1;
b_fir_2 = ones(1, M_2) / M_2;

delta_fir_1 = sqrt(filter(b_fir_1, 1, aud_squared));
delta_fir_2 = sqrt(filter(b_fir_2, 1, aud_squared));

gain_equalized_fir_1 = aud ./ delta_fir_1;
gain_equalized_fir_2 = aud ./ delta_fir_2;
```

```
figure(4)
subplot(2,1,1);
plot(aud_squared(2700:6700)); hold on;
plot(delta_fir_1(2700:6700), 'r'); hold off;
title('S5 Squared Signal and FIR Delta 1');
xlabel('Sample Number'); ylabel('Amplitude');
grid on;

subplot(2,1,2);
plot(gain_equalized_fir_1(2700:6700));
title('S5 Gain Equalized FIR Signal 1');
xlabel('Sample Number'); ylabel('Amplitude');
grid on;

...
```

- **Results and Analysis:**

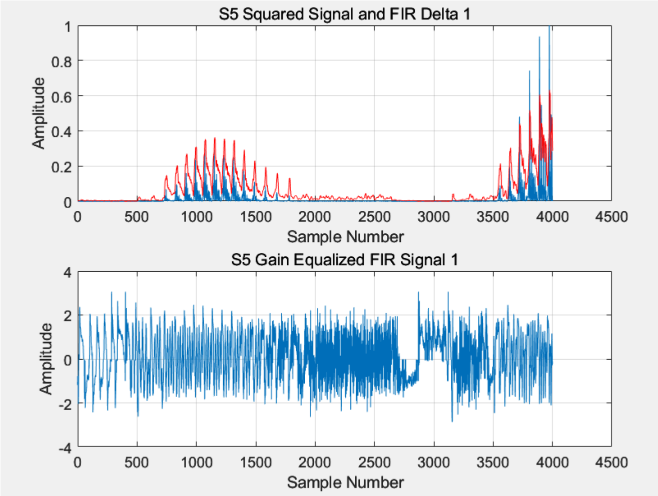


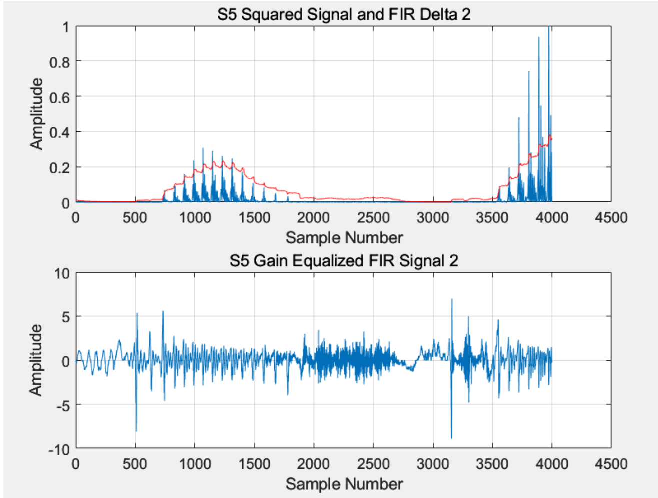




Delta 1 corresponds to alpha=0.9, and delta 2 corresponds to alpha=0.99.

Here alpha=0.9 have a larger rate of adapting to the changing speech level. The overall level of the signal is equalize to approximately the same, although in x=2800 there are some abnormalities. On the contrary, the equalizer using alpha=0.99 mostly does compression for large signal levels, and is less sensitive to rapid changes.





The results using FIR is mostly the same, with smaller M corresponding to smaller alpha.

Also for Delta 2, the deviation is more stable for both the overall waveform and the near-zero initial values. Also, the equalized FIR signal has a bigger amplitude than the IIR equalized signal.

Conclusion

In this lab, we explored μ-law quantization and adaptive quantization techniques for speech signals.

- **μ-Law Quantization**: The process significantly increases the amplitude of small signals, improving encoding accuracy. Our experiments showed that the quantization error decreases as the number of bits increases, with 10-bit quantization closely resembling white noise. The inverse μ-law function was validated, showing minimal reconstruction error.
- **SNR Analysis**: Comparing uniform and μ-law quantization methods, it was observed that for every additional bit in uniform quantization, the SNR improved by approximately 6 dB. In contrast, μ-law quantization maintained relatively stable SNR values across varying signal levels, achieving optimal performance at μ=255, a standard parameter for telephone speech coding.
- **Adaptive Quantization**: Using both IIR and FIR filters to dynamically adjust quantization based on signal characteristics demonstrated that smaller α values (IIR) or M values (FIR) allowed faster adaptation to changes in speech levels. However, larger α or M values provided more stable deviation handling but were less responsive to rapid changes.

In summary, μ-law quantization offers significant advantages over static uniform quantization in terms of dynamic range and fidelity, especially for low-amplitude signals. Adaptive quantization techniques further enhance these benefits by adjusting to signal dynamics, ensuring better performance in real-world applications.