



Southern University of Science and Technology

Speech Signal Processing

## **Lab 9 Report**

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## Question 11.27

(a)

Code:

```
x = -1:0.001:1;
y = zeros(6,length(x));
mu = [1, 20, 50, 100, 255, 500];
figure;hold on;
for i = 1:6
    y(i,:) = mulaw(x, mu(i));
    plot(x,y(i,:));
end
```

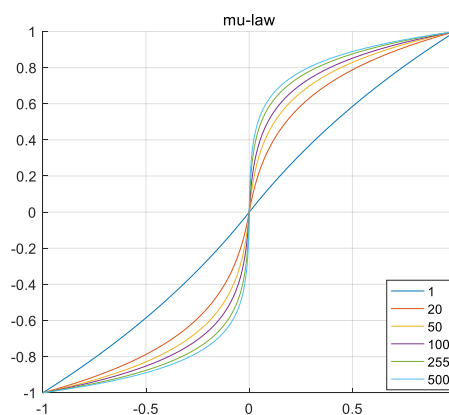


Figure 1

(b)

```
clear;clc
signal = audioread('s5.wav');
signal = signal(1300:18800);
y = mulaw(signal, 255);
```

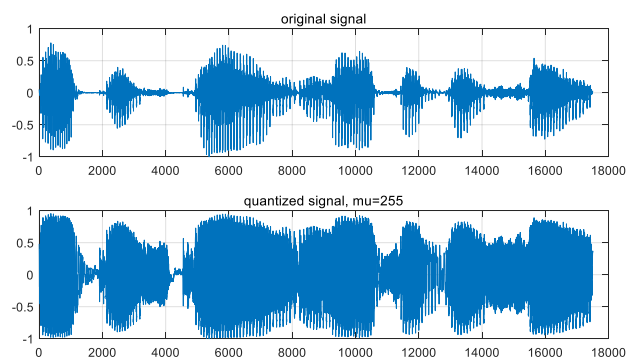


Figure 2

As we can see in figure 1, the mu-law can amplify small amplitude components, while holding the great amplitude components, which could be regarded as augmenting the details.

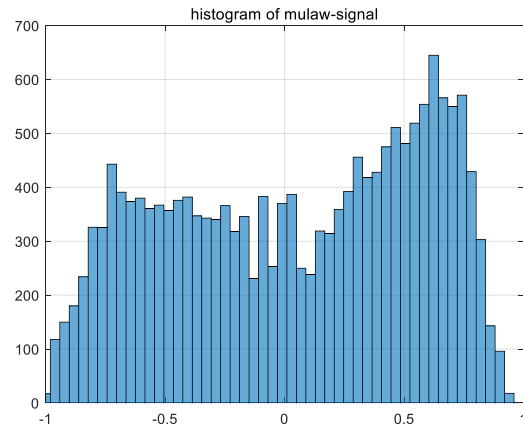


Figure 3

While the input signal has a Laplace distributed histogram, the expanded signal has a wider histogram. Even though, it is better than the uniform quantization.

(c)

The inverse mu-law function has been implemented already. Here I just show the result and do a subtraction with the original signal:

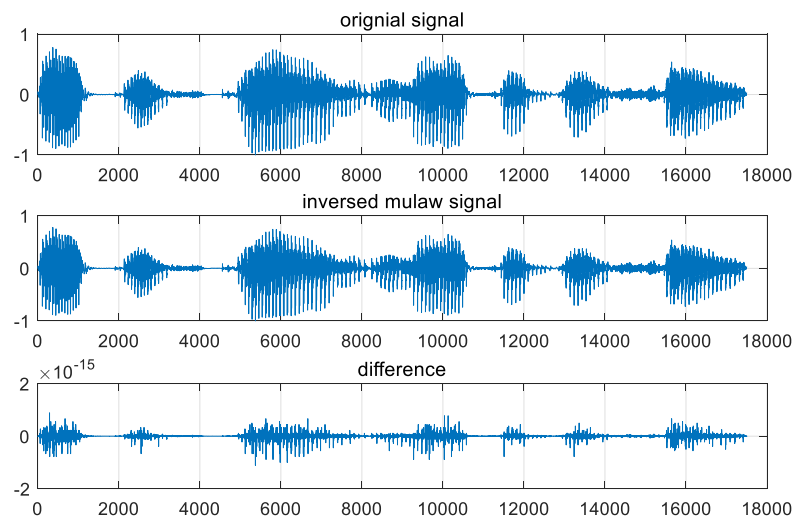


Figure 4

The difference is at a negligible scale, which I think is due to the computation error. It is confident to say that the inversed signal is identical to the original one.

(d)

Code:

```
clear;clc
signal = audioread('s5.wav');
signal = signal(1300:18800);
bit = [4,6,8,10];
yh = zeros(4,length(signal));
e = zeros(4,length(signal));
for i = 1:4
    yh(i,:) = fxquant(mulaw(signal,255),bit(i),'round','sat');
    e(i,:) = mulawinv(yh(i,:),255)-signal;
    error = e(i,:);

    subplot(2,2,i);histogram(error(1:8000),50);title(sprintf('bit
    = %d',bit(i)));grid;
end
figure;hold on;
for i = 1:4
    error = e(i,:);
    pspect(error(1:8000), 8000, 1024, 60);
end
title('error power spectrum'); legend('bit=4', 'bit=6',
'bit=8', 'bit=10');grid;
```

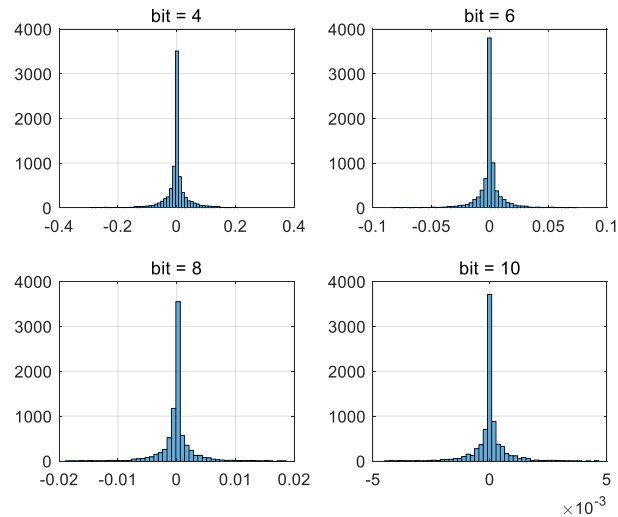


Figure 5, error signal histogram

After companding manipulation, the error signal has a similar histogram with the input signal, and has more zero values. When the bit number increases, the distribution of error signal remains the same, but the scale has been significantly reduced. This is identical with the previous conclusion that increasing bit number results to increasing SNR.

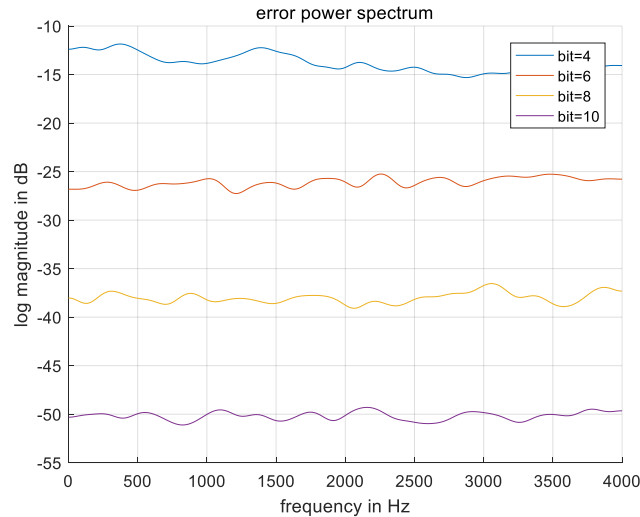


Figure 6

When observing the power spectrum, the conclusion is clearer to see. As discussed on class, increasing 1 bit can increase the SNR by 6 dB. Here we can see that the noise signal has a uniform distribution along frequencies, and decrease by roughly 12 dB for every 2 bits.

## Question 11.29

Code:

```
signal = audioread('s5.wav');
signal = signal';
sigma = zeros(1,length(signal));
alpha = 0.9; % run for 0.9 and 0.99
for i = 2:length(sigma)
    sigma(i) = alpha*sigma(i-1)+(1-alpha)*(signal(i))^2;
end
sigma = sigma.^(0.5);
subplot(2,1,1); hold on;
plot(2700:6700,signal(2700:6700));plot(2700:6700,sigma(2700:6700));
legend('speech', 'standard deviation');title('alpha = 0.99');
hold off;
subplot(2,1,2);plot(2700:6700,signal(2700:6700)./sigma(2700:6700));
title('gain equalized speech');
```

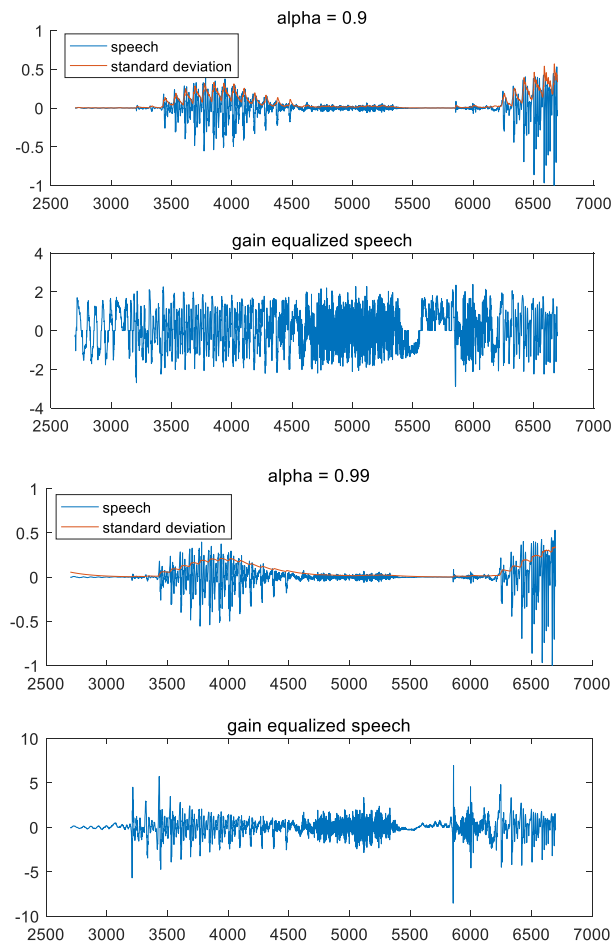


Figure 7, Gain adaptive

When  $\alpha = 0.9$ , the standard deviation adjusts rapidly, results in a more uniform equalized speech. However, the originally small noise signal has also been hugely amplified.

When  $\alpha = 0.99$ , the deviation has a smoother curve, thus the result isn't rapidly equalized. The dynamic range is also wider than the previous one. Another advantage is that the noise signal is not hugely amplified.

(b)

Code:

```
signal = audioread('s5.wav');
signal = signal';
sigma = zeros(1,length(signal));
M = 10;
for i = M+1:length(sigma)
    sigma(i) = sum(signal(i-M+1:i).^2);
end
sigma = sigma.^(0.5);
subplot(2,1,1);
plot(2700:6700,signal(2700:6700)./sigma(2700:6700));
title('M=10');
M = 100;
```

```

for i = M+1:length(sigma)
    sigma(i) = sum(signal(i-M+1:i).^2);
end
sigma = sigma.^(0.5);
subplot(2,1,2);plot(2700:6700,signal(2700:6700)./sigma(2700:6700));
title('M=100');

```

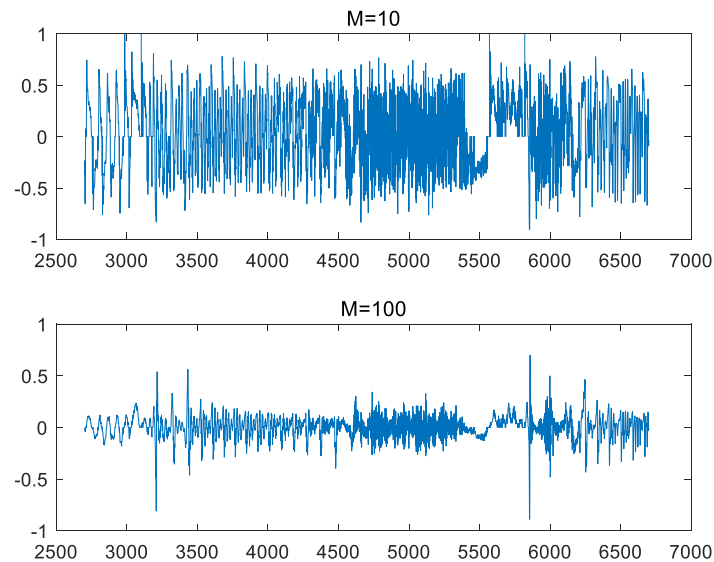


Figure 8, FIR implementation

For  $M = 10$ , the result is like  $\alpha = 0.9$ , since a smaller window can provide a faster adaptation, just as when  $x[n]$  presents fewer weight in previous IIR method.

For  $M = 100$ , it is similar with  $\alpha = 0.99$ , since a bigger window means slower adaptation. However, the noise amplification is more significant than IIR method.