Southern University of Science and Technology

Speech Signal Processing

Lab 8 Report

11510478 郭锦岳

Question 11.24

(a)

Code:

```
signal = [audioread('s1.wav')', audioread('s2.wav')',
audioread('s3.wav')', audioread('s4.wav')',
audioread('s5.wav')',audioread('s6.wav')']';
min = min(signal);
max = max(signal);
mean = mean(signal);
variance = var(signal);
stem((min:(max-
min)/24:max),hist(signal,25),'Linewidth',10,'Marker','non
e');
stem((min:(max-
min)/99:max),hist(signal,100),'Linewidth',2.5,'Marker','n
one');
```

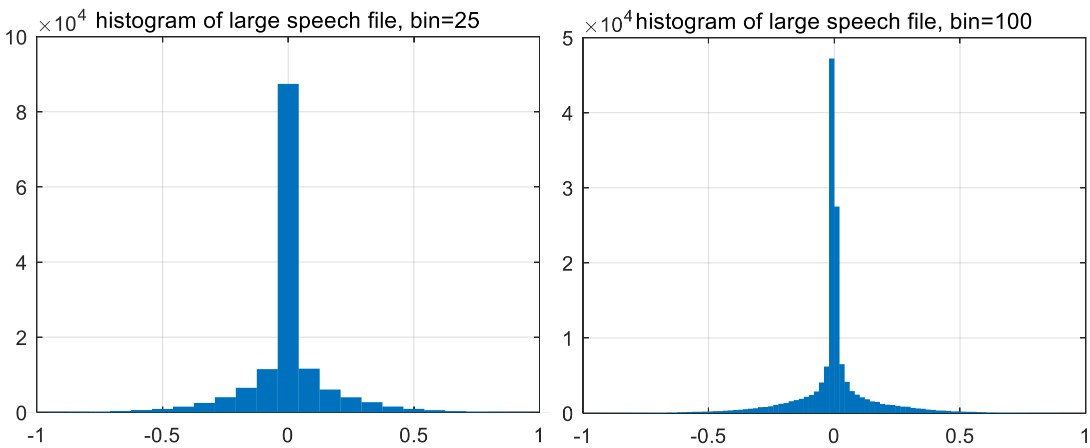

Figure 1, Speech file histograms

With more bins, the histogram demonstrates a more significant centering phenomenon, showing that the speech signal has a density distribution like laplacian distribution.

(b)

Code:

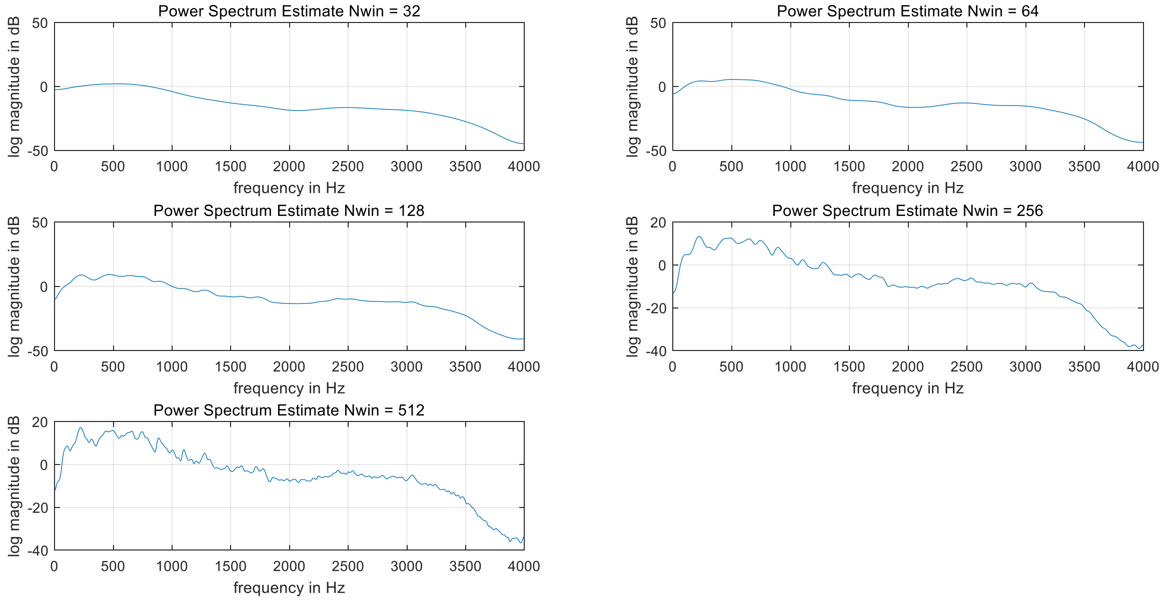

Figure 2, Power spectra with different window lengths

With a greater window length, the power spectrum of the speech signal shows more high frequency components, which is because the property of Fourier Transform. Besides, the amplitude of the power spectrum is also greater.

(d)

Code:

```
male = [audioread('s2.wav')', audioread('s4.wav')',
audioread('s5.wav')', audioread('s6.wav')']';
female = [audioread('s1.wav')', audioread('s3.wav')']';
hold on; pspect(male, 8000, 1024, 256); pspect(female, 8000,
1024,256); hold off;grid;
```

Figure 3, power spectra of two signals with different window lengths

As a different result of the slides, the female voice has a greater amplitude at low frequency. However, when increase the window length to 256, the amplitude doesn't show a significant greater amplitude at low frequency. Besides, the male voice has a peak at 2500H Hz, while female voice has a peak at 3000 Hz.

Question 11.25

(a)

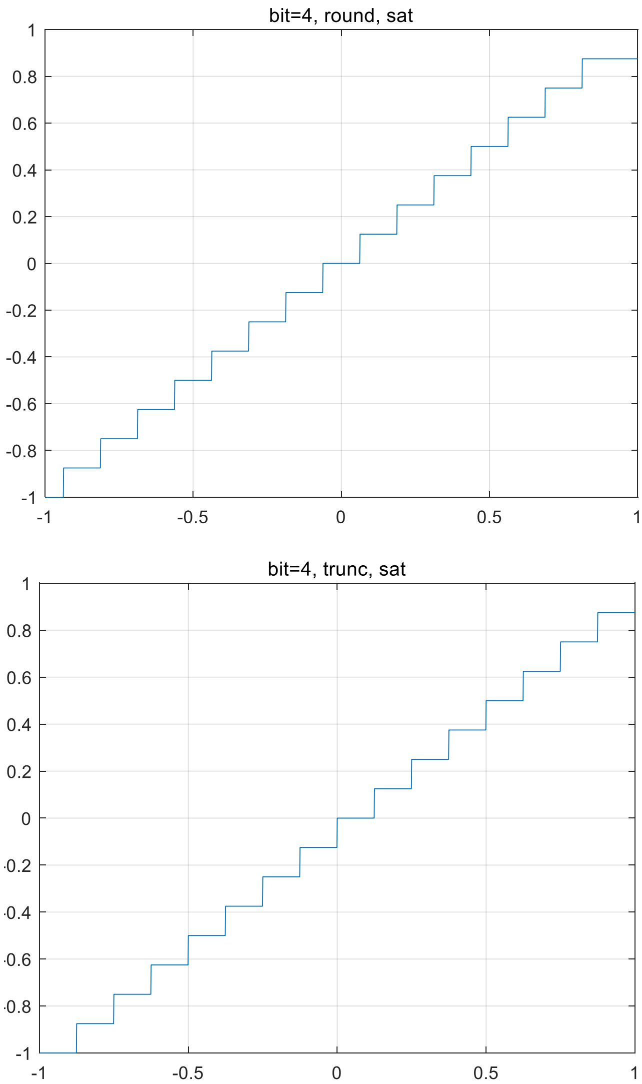

Figure 4, Quantizers

The round quantizer is identical with mid-tread quantizer on the slides, while trunc quantizer is a little bit different with mid-riser quantizer.

The range of quantization error $e[n]$ is $[-\Delta/2, \Delta/2]$ for round quantizer, and $[\Delta, 0]$ for trunc quantizer.

(b)

Code:

```
signal = audioread('s5.wav')';
signal = signal(1300:18800);
signal_quant = zeros(3,length(signal));
bit = [10, 8, 4];
% quantization period
figure;
subplot(4,1,1);plot(signal);title('original signal');
for i = 1:3
    signal_quant(i,:) = fxquant(signal, bit(i), 'round',
'sat');

subplot(4,1,i+1);plot(signal_quant(i,:));title(sprintf('quanti
zed signal with %d bits', bit(i)));
end
```

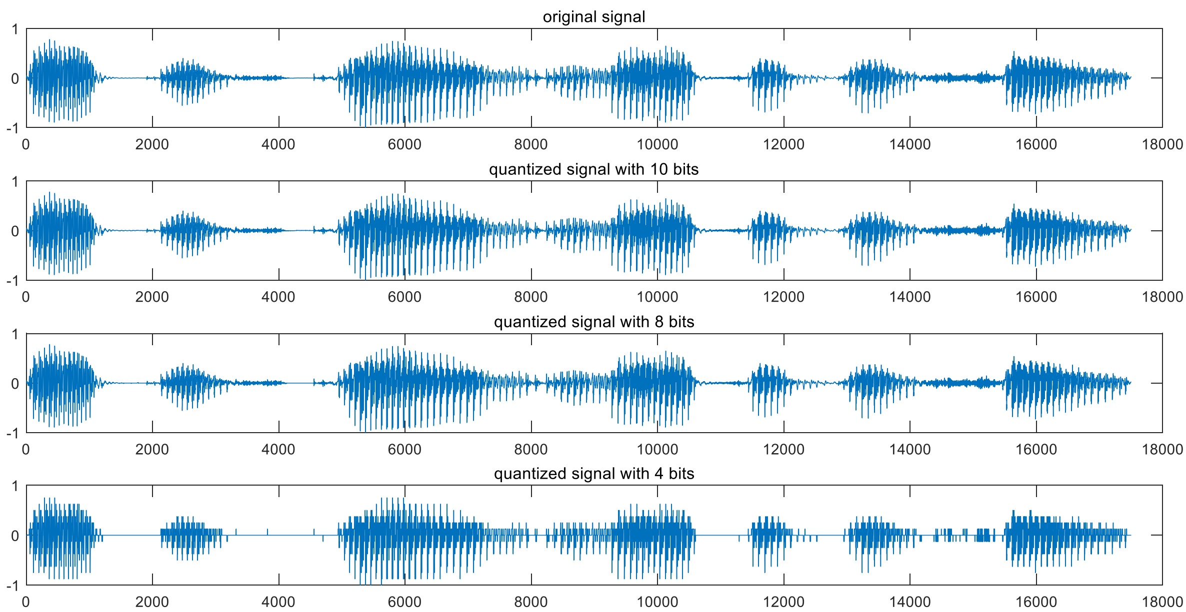

Figure 5, Quantized signal

```
% compute error
signal_error = zeros(3,length(signal));
for i = 1:3
    signal_error(i,:) = signal_quant(i,:) - signal;
    error = signal_error(i,:);
    figure;
    striplot(error(1:8000), 8000,
2000);title(sprintf('error sequence , bit = %d', bit(i)));
end
```

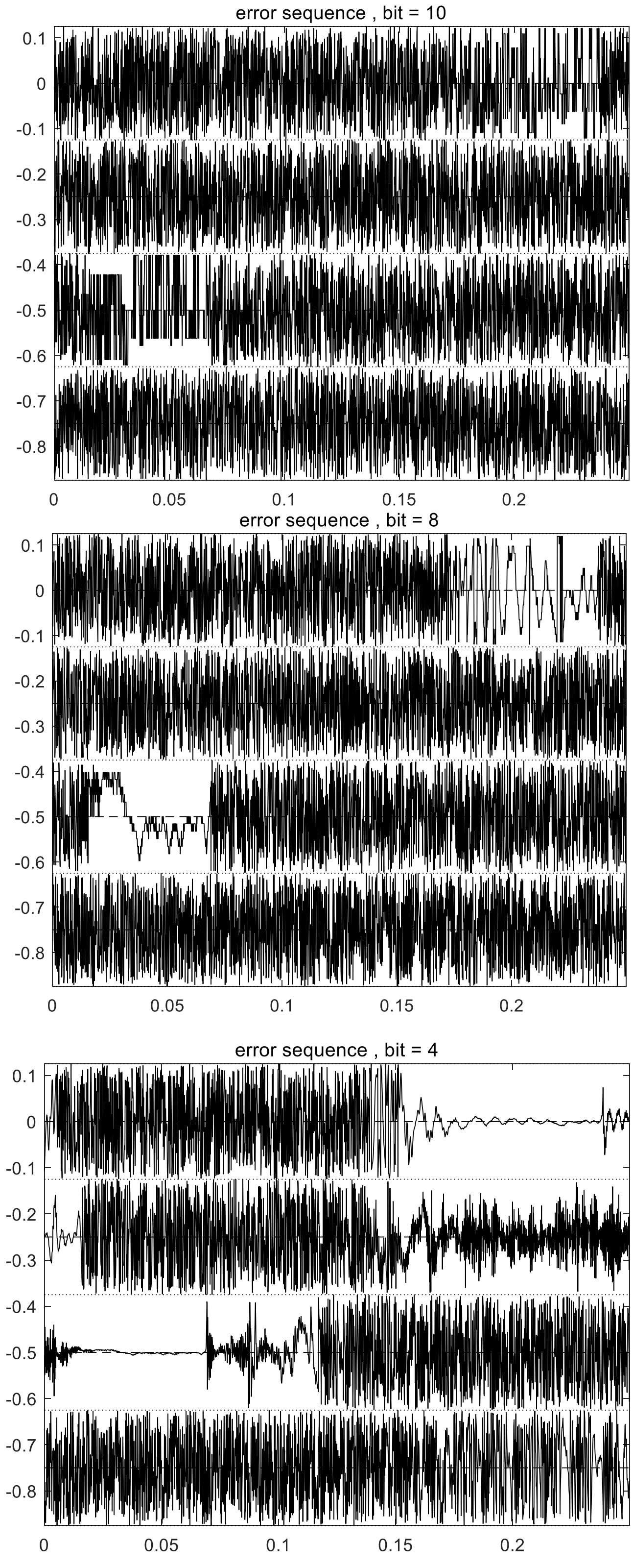

Figure 6, error sequences of different bit numbers.

For 10-bit results, it shows an almost perfect reconstruction; But for lower bit numbers, the signal shows distortion, while the error sequence might be identical with the signal. This is because the amplitude is smaller than the minimum step size of the quantizer, thus the whole signal becomes error.

Therefore, sufficient bit number is necessary to avoid huge quantization error.

(c)

Code:

```
figure;
hold on;pspect(signal, 8000, 1024, 128);
for i = 1:3
    signal_error(i,:) = signal_quant(i,:) - signal;
    error = signal_error(i,:);
    pspect(error, 8000, 1024, 128);title('error signal power
spectra');
end
hold off;
```

Figure 7, error signal power spectra

As a theory of thumb, the greater bit number shows a lower quantization error. According to the slides, we can see the quantization error as a gaussian white distributed error. And when increase for one bit, the SNR will increase for 6dB. We can see on the amplitude of the error signal that its amplitude decreases by 6 dB for every additional bit.