Lab 3

Sample Rate Alteration

0. Abstract

Sample Rate Alteration is the process of changing the sampling rate or sampling frequency of a discrete signal to obtain a new discrete representation of the underlying continuous signal. In this lab, we will mainly using downsample and upsample to the original signal and see what will change when comparing to the original signal.

1. Sections

1.1. Sample the signal and downsample it

The code for the is as follows:
```
function exe1_1()
f = 10;
t = 0 : 0.0001 : 1;
L = 1000;
x = cos(2 * pi * f * t);
xd = x ( 1 : L : end )
figure(2)
stem(xd)
```

Because it is first sampled at 10k Hz and then selecting every L sample from the x[n] starting with the first one and the downsampled signal has a sampleing rate of 10/L kHz.

1.2. Plot the spectrum of the original signal

The code is as follows:
```
function exe1_2()
f = 1000;
t = 0 : 0.00001 : 1;
x = cos(2 * pi * f * t);
y = fft(x)
stem(y)
```

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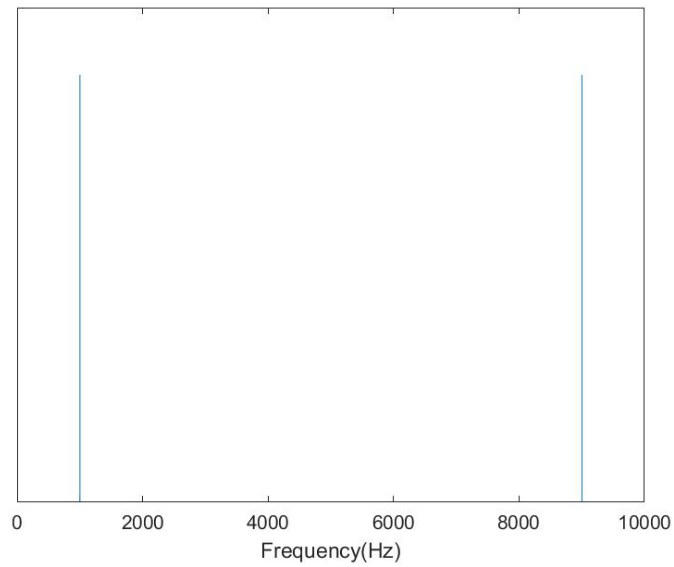

Figure 1: The spectrum of the original signal

1.3. Downsample the signal by factors L

The code is as follows:

```
function exe1_3()
f = 1000;
t = 0 : 0.0001 : 1;
L = [1 2 4 8];
x = cos(2 * pi * f * t);
xd1 = x ( 1 : L(1) : end )
xd2 = x ( 1 : L(2) : end )
xd3 = x ( 1 : L(3) : end )
xd4 = x ( 1 : L(4) : end )
subplot(2,2,1);stem(fft(xd1),'Marker','none');set(gca,'ytick',t);xlabel('Frequ
ency(Hz)');title('L = 1')
subplot(2,2,2);stem(fft(xd2),'Marker','none');set(gca,'ytick',t);xlabel('Frequ
ency(Hz)');title('L = 2')
subplot(2,2,3);stem(fft(xd3),'Marker','none');set(gca,'ytick',t);xlabel('Frequ
ency(Hz)');title('L = 4')
subplot(2,2,4);stem(fft(xd4),'Marker','none');set(gca,'ytick',t);xlabel('Frequ
ency(Hz)');title('L = 8')
```

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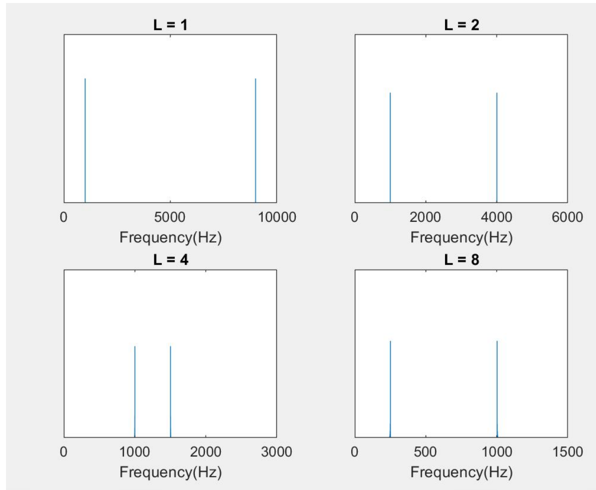

Figure 2: Downsample the signal by L = 2, 4, 8

1.4. Explanation when L = 8

When L = 8, since the original sample rate is 10k Hz and the formula is 10k/L Hz, the frequency of the downsample tone is 10k ./ 8 = 1250 Hz.

1.5. Load and Plot the image from the image.zip file

The comments for the code are as follow:

```
function exe1_5()
y = imread('barbaraLarge.gif'); % import the image
L = 2; % set the value of L
[rows,columns] = size(y); % get the size of rows and columns
yd = zeros(floor(rows/L),floor(columns/L)); % create the vector
for i=1:floor(rows/L) % do the downsample by rows
for j=1:floor(columns/L) %do the downsample by columns
yd(i,j) = y((i-1)*L+1, (j-1)*L+1);
end
end
figure('Position',[100 100 size(y,2) size(y,1)], ...% some settings for the
figure
'MenuBar', 'none', 'PaperPosition',[0.25 1.5 5 5]);
axes('Position',[0 0 1 1]); %set the axis of x and y
imagesc(yd); %show the image
colormap('gray');
```

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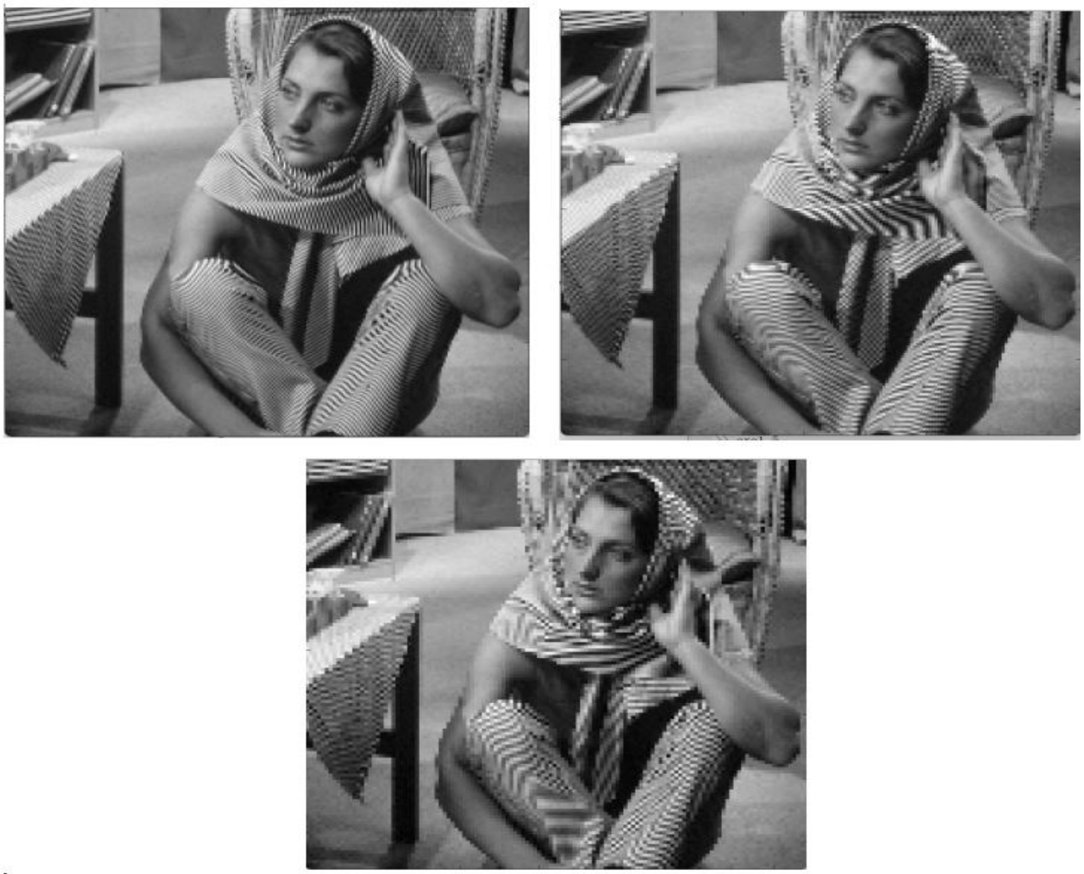

Figure 3: Downsample the image by factors of L = 2, 3, and 4 respectively

1.6. Make some observations of the downsampled images

In the original image is stripe is straight but the stripes become to bend as the L increases. The griddings on the rattan chair become blurred and her expression is more difficult to tell. Something like mosaic appears on the book as L increases.

The printers can not downsample the image too much which means they can not increases the L too much, because it will cause the straight stripes to bend.

1.7 Calculate the downsample factors

The integer downsample factor is 5 and calculation is as follows:
44.1 ./ 9

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ans =

4.9000

1.8 Display the sonogram of this downsampled signal

The code is as follows:
```
function exe1_8()
[x,Fs]=audioread('speech_female.wav');
x = x(:,1);
x = x';
N = length(x);
t = (0:N-1)/Fs;
y = fft(x);
f = Fs/N*(0:round(N/2)-1);
figure(1)
subplot(211);
plot(t(1:Fs * 1.4),x(1:Fs * 1.4),'g');
xlabel('Time/s');ylabel('Amplitude');
title('Original Signal');
grid;
subplot(212);
plot(t(1:5:Fs * 1.4),x(1:5:Fs*1.4),'g');
xlabel('Time/s');ylabel('Amplitude');
title('Downsample Signal');
grid;
figure(2)
y = fft(x(1:Fs * 1.4));
y1 = fft(x(1:5:Fs * 1.4));
subplot(211); plot(abs(y));title('Original Signal');axis([0 4410 0 1000]);
subplot(212); plot(abs(y1));title('Original Signal');axis([0 4410 0 1000]);
```

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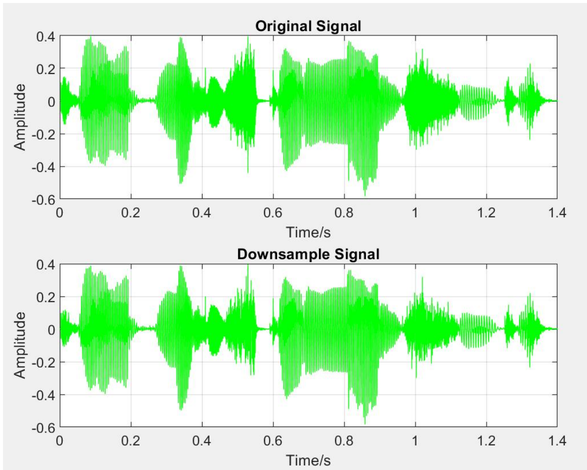

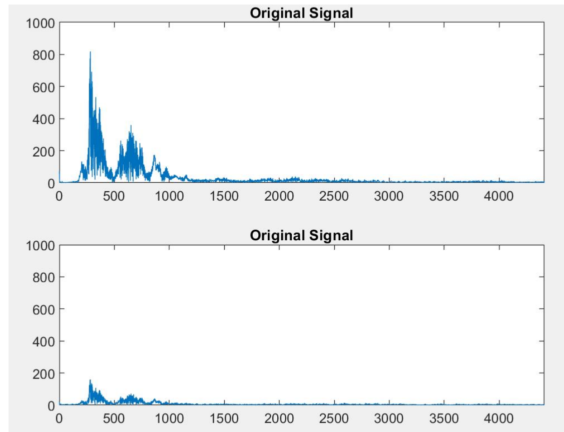

Figure 4 & 5:The sonogram of the original signal and the downsample signal

1.9 Describe the differences between the two spectrograms

In general, the downsample signal loses some energy comparing to the original signal. The energy is missing in the downsamped signal where there are relatively large fluctuations in the amplitude. Since the skipped sample will cause the energy lose. That lady is speaking the word “medicine” when around 1 second. The sonogram is very dense around 1 second in original signal. But it is sparse in the downsmapled signal.

1.10 Listen to the entirety of the two sounds

The sound of the original signal is very loud and clear. While there is some kind of noise on the end of words and the sound is very weak.

1.11 Display the spectrogram of this filtered and downsampled signal

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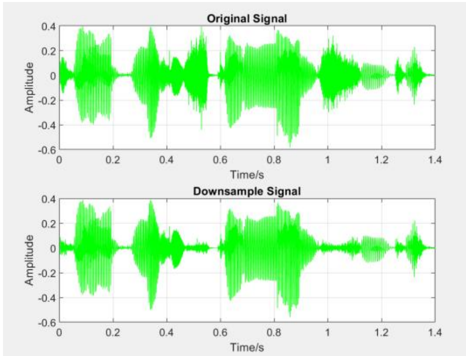

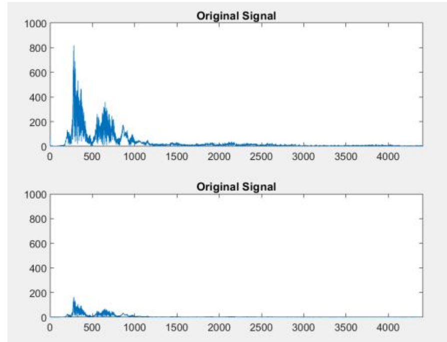

Figure 6 & 7: The spectrogram of the filtered and downsampled signal and the original signal

1.12 Display the spectrogram of this filtered and downsampled signal

I the spectrogram in 1.8 the amplitude at 1 second is very large but in spectrogram in 1.12 the amplitude is very small.

1.13 Listen to the results of downsampling

The sound is still weak when using the lowpass filter, but the noise at the end of each word is filtered. The results with prior lowpass is more listenable.

1.14 Ways to administered medicine to animals

I used to make the pill crush into powder and mix it with food.

2.1 Sample the signal

The code is as follows:

```
function exe2_1()
t = 0 : 0.001 : 1;
x = cos(2 * pi * f * t);
y = upsample(x,M)
```

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2.2 Upsample a sampled sinusoid by M

The code is as follows:

```
function exe2_2()
t = 0 : 0.001 : 1;
Fs = 1000;
f = 200;
x = cos(2 * pi * f * t);
M = [1 2 4 8]
for i = 1:length(M)
subplot(2,2,i);plot(abs(fft(upsample(x,M(i)))));set(gca,'ytick',t);title('L =
' ,M(i));axis([0 (M(i)*Fs) 0 10]);xlabel('Frequency(Hz)')
end
```

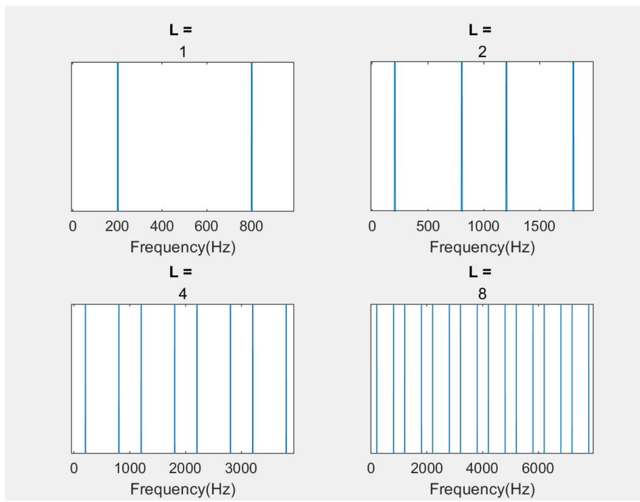

Figure 8: The spectrogram of the signal created in 1.1, with f = 200Hz

2.3 Observation on the spectrogram

Because when we multiply the impulse respnse of the lowpss FIR filter by a series of alternating positive and negative 1's. And since the impulse response is symmetric and the value on the even size is zero and the only response it changes is h[6]. Because it turns from the positive to negative, it causes the filter to be a high pass filter.

When we upsample the signal higher, more frequencies will be discovered and the component at f = 200Hz remain in all of them.

2.4 Plot the resulting upsampled and filtered signal

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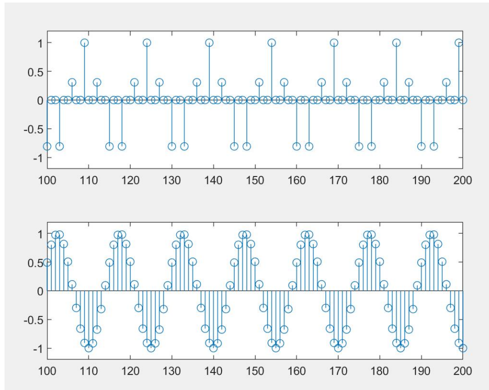

Figure 8: The spectrogram of the signal created in 1.1, with f = 200Hz

2. Conclusion

In lab3, we get start with the downsample and analyzing the energy loss comparing to the original signal. We also apply a lowpass filter to the signal and it help to make the sound clearer. And then we move on to the upsample and apply a lowpass filter to it . Application areas include image scaling[2] and audio/visual systems, where different sampling rates may be used for engineering, economic, or historical reasons.

3. Acknowledgments

Thanks for the TA gives the instruction on Lab3.

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