EE 542

Applications
In
Digital Signal Processing

Computer Assignment #3

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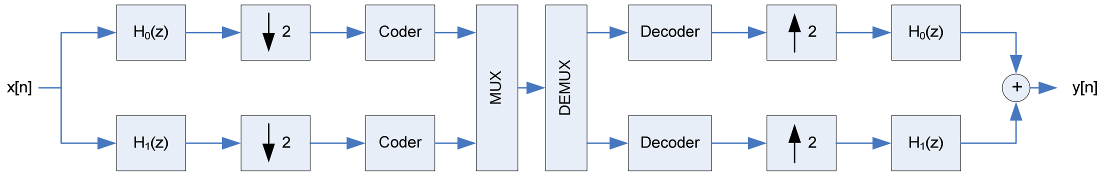
1. Introduction

Sub-band Coding. In this project you are expected to design a sub-band coder/decoder for a speech signal. The structure of your analysis/synthesis filter bank should have the basic unit which appears as in figure below. The final filter bank structure should consist of at least a two-level tree structure as shown in Figure 14.30 (page 837).



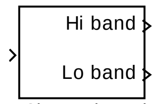
You should make an appropriate choice of filters $h_i[n]$ and $g_i[n]$, and should be consistent with perfect reconstruction and alias-free conditions.

Assume the coder/decoder to be a quantization operation, with the option that each sub-band component can have a different quantization level. You can use heuristics, to determine the number of bits that would be allocated for each sub-band component. Explain your choice.

2. Simulation & Results

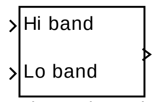
2.1 Simulation version #1

In this version, a Two-Channel Analysis Sub-band Filter and a Two-Channel Synthesis Sub-band Filter was used.



Two-Channel Analysis Subband Filter

The Analysis Sub-band Filter decomposes a signal into a high-frequency sub-band (High band) and a low-frequency sub-band (Low band) using the specified highpass and lowpass FIR filters. Each sub-band has half the bandwidth and half the sample rate of the original signal. Usually, the highpass and lowpass filters should be half-band filters designed to complement each other. This block accepts sample- and frame-based inputs of all sizes.

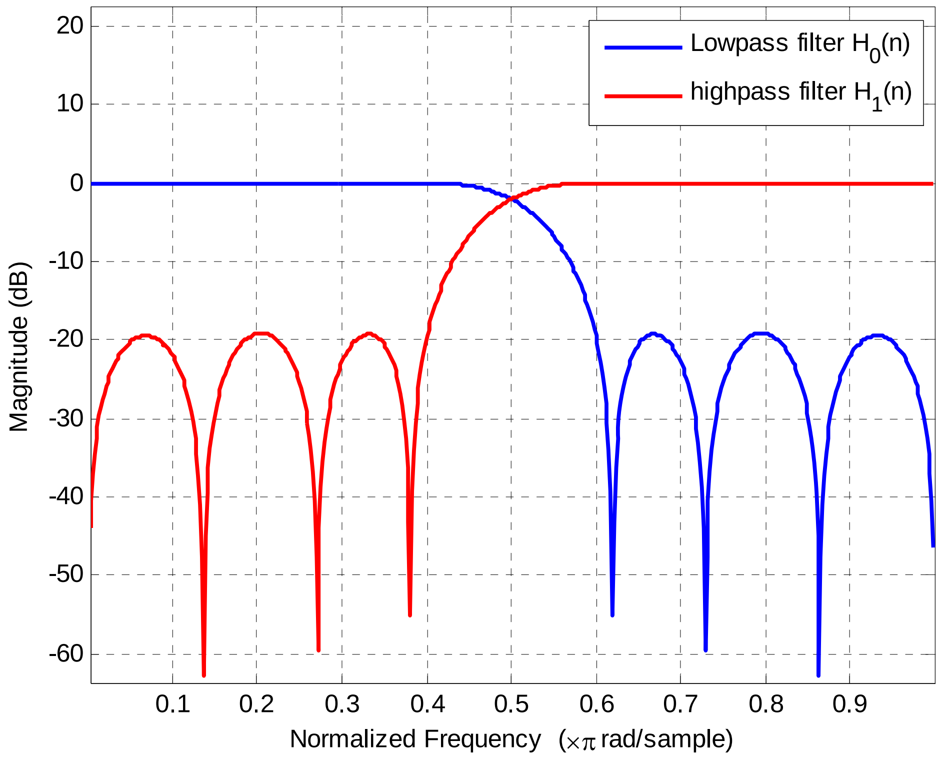


Two-Channel Synthesis Subband Filter

The Synthesis Sub-band Filter reconstruct a signal from a high-frequency sub-band (High band) and a low-frequency sub-band (Low band) using the specified highpass and lowpass FIR filters. The input subbands should have the same bandwidths and sample rates. Usually, the highpass and lowpass filters should be half-band filters designed to complement each other. This block accepts sample- and frame-based inputs of all sizes.

Table 2-1 shows the coefficients of the filters I used.

The following figures show the frequency response of these filters.


Lowpass filter $H_0(n)$
highpass filter $H_1(n)$
Magnitude (dB)
20
10
0
-10
-20
-30
-40
-50
-60
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
Normalized Frequency ($\times\pi$ rad/sample)

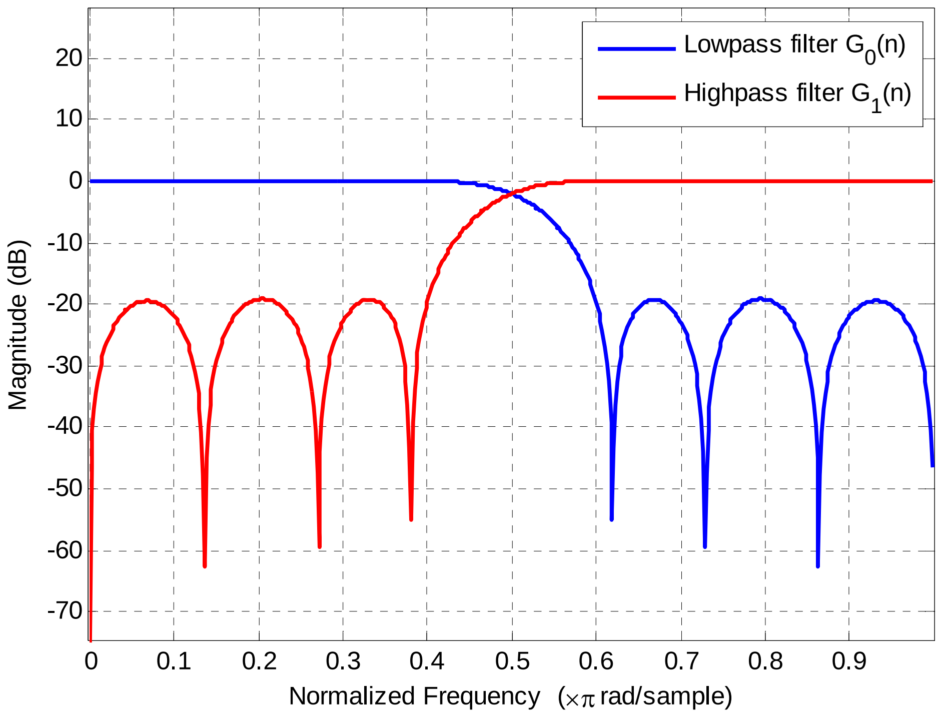

Lowpass filter $G_0(n)$
Highpass filter $G_1(n)$
Magnitude (dB)
20
10
0
-10
-20
-30
-40
-50
-60
-70
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
Normalized Frequency ($\times\pi$ rad/sample)

Table 2-1. Filter Coefficients.

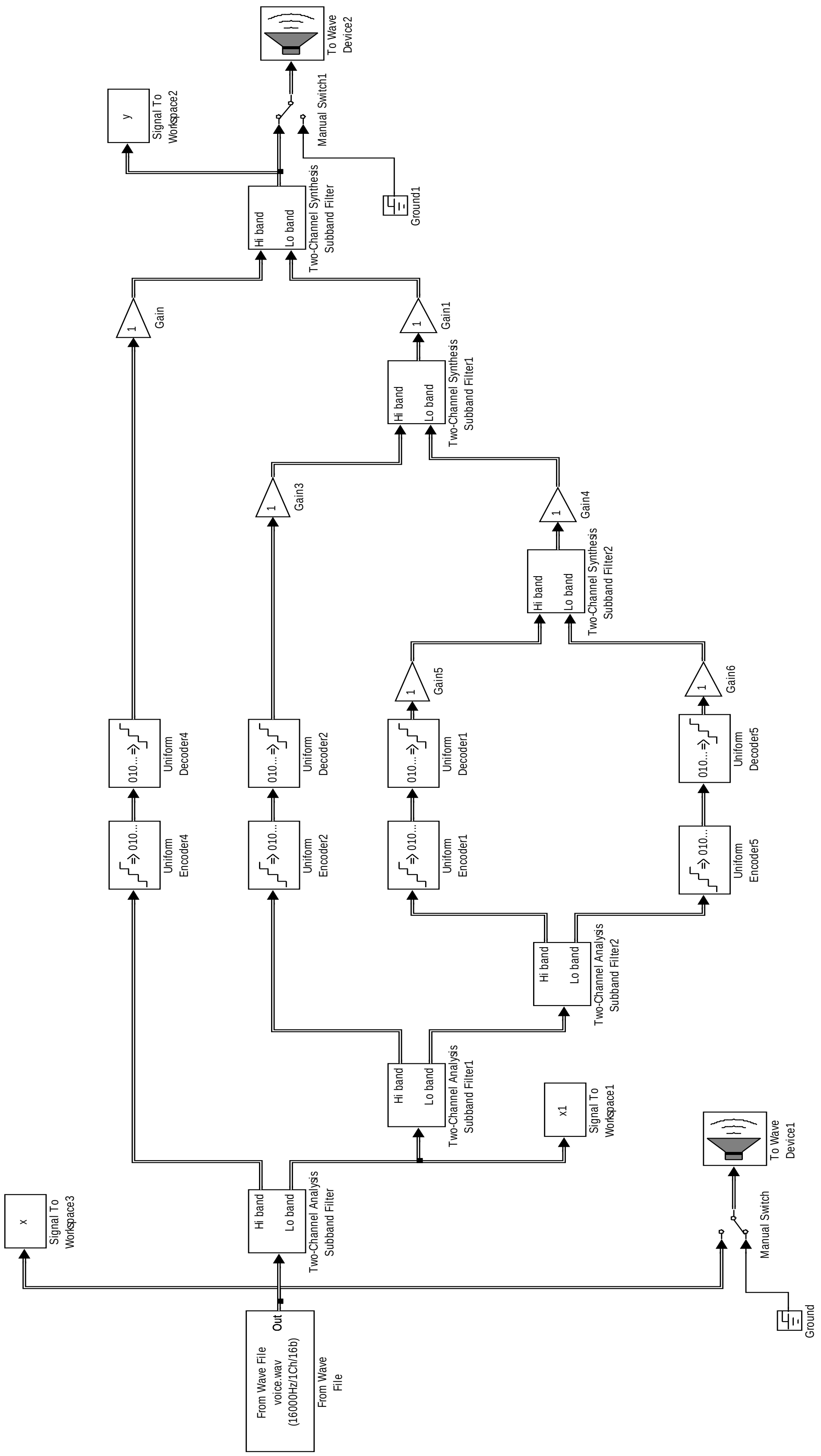
n	$H_0(n)$	$H_1(n)$	$G_0(n)$	$G_1(n)$
1	0.281390464166301	0.281390464166301	0.000760751201581	-0.000760751201581
2	0.508284208701808	-0.508284208701808	-0.012090482145191	-0.012090482145191
3	0.357732622230456	0.357732622230456	0.033404904118193	-0.033404904118193
4	-0.038190157723878	0.038190157723878	-0.031567481307907	-0.031567481307907
5	-0.175603688788674	-0.175603688788674	-0.014024436478079	0.014024436478079
6	0.008622989460103	-0.008622989460103	0.042528889784377	0.042528889784377
7	0.104814801604155	0.104814801604155	0.011662785870338	-0.011662785870338
8	-0.009211577871239	0.009211577871239	-0.06589565826469	-0.06589565826469
9	-0.06589565826469	-0.06589565826469	-0.009211577871239	0.009211577871239
10	0.011662785870338	-0.011662785870338	0.104814801604155	0.104814801604155
11	0.042528889784377	0.042528889784377	0.008622989460103	-0.008622989460103
12	-0.014024436478079	0.014024436478079	-0.175603688788674	-0.175603688788674
13	-0.031567481307907	-0.031567481307907	-0.038190157723878	0.038190157723878
14	0.033404904118193	-0.033404904118193	0.357732622230456	0.357732622230456
15	-0.012090482145191	-0.012090482145191	0.508284208701808	-0.508284208701808
16	0.000760751201581	-0.000760751201581	0.281390464166301	0.281390464166301

The figure in next page is the structure of this sub-band coding system. Note that the up-sample and down-sample parts have already been included inside those analysis and synthesis filters.

This is a three level filter bank structure consists of four pair of encoder/decoders which were assumed be the quantization operation. The Encoder uniformly quantizes and encodes the input into specified number of bits. The input is saturated at positive and negative peak value. The decoder uniformly decodes the input with positive and negative peak value.

The input signal x[n] is a 16000 Hz sampling rate, one channel and 16-bit human voice signal containing 45312 samples. And the output signal is y[n].

I allocated different sets of bits for each sub-band component. The following results are the SNR versus these different sets of bits. “(H)” and “(L)” mean high band and low band in the structure.


From Wave File
voice.wav
(16000Hz/1Ch/16b)
Out
From Wave File
Signal To Workspace3
x
Two-Channel Analysis Subband Filter
Hi band
Lo band
Two-Channel Analysis Subband Filter1
Two-Channel Analysis Subband Filter2
Signal To Workspace1
x1
Uniform Encoder4
Uniform Encoder2
Uniform Encoder1
Uniform Encoder5
Uniform Decoder4
Uniform Decoder2
Uniform Decoder1
Uniform Decoder5
010...
Gain
Gain1
Gain3
Gain4
Gain5
Gain6
1
Two-Channel Synthesis Subband Filter
Two-Channel Synthesis Subband Filter1
Two-Channel Synthesis Subband Filter2
Signal To Workspace2
y
Manual Switch1
Manual Switch
Ground1
Ground
To Wave Device2
To Wave Device1

Sub-band encoder/decoder	SNR	Frequency content of input signal	Frequency content of output signal
16(L)+16+16+16(H)	18.6850 dB	Fig 2.4	Fig 2.5
16(L)+16+16+8(H)	18.6218 dB		Fig 2.7
16(L)+16+8+8(H)	18.5899 dB		Fig 2.9
16(L)+8+8+8(H)	18.5619 dB		Fig 2.11
8(L)+8+8+8(H)	18.0083 dB		
8(L)+8+8+4(H)	-10.9911 dB		Fig 2.6
8(L)+8+4+4(H)	-13.3697 dB		Fig 2.8
8(L)+4+4+4(H)	-14.7300 dB		Fig 2.10
4(L)+4+4+4(H)	-21.8679 dB		Fig 2.12

Fig 2.4 Frequency content of input signal

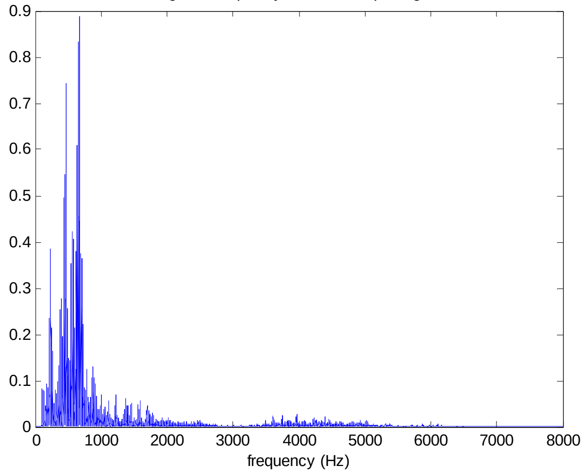


Fig 2.5 Frequency content of output signal

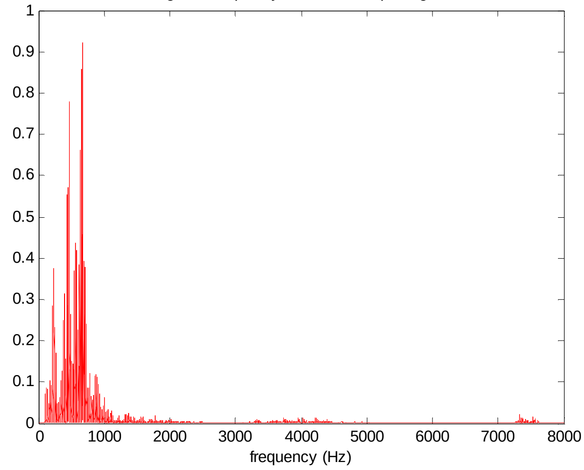


Fig 2.7 Frequency content of output signal

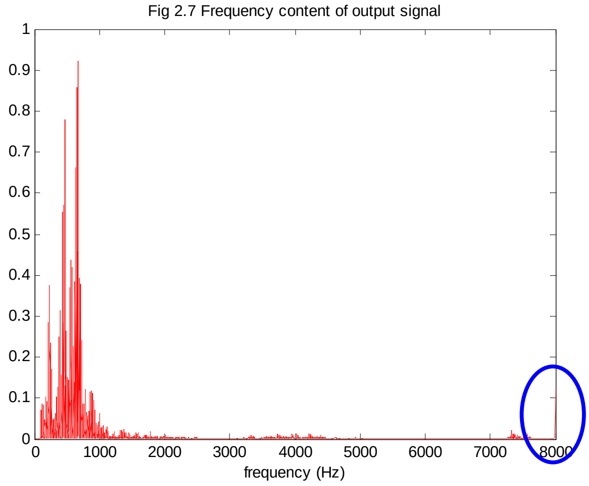


Fig 2.9 Frequency content of output signal

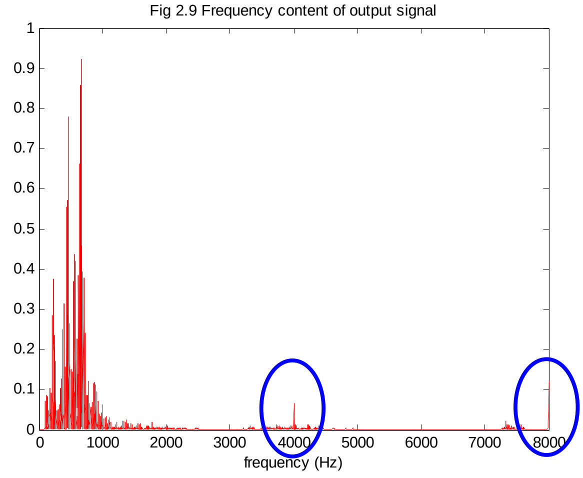


Fig 2.11 Frequency content of output signal

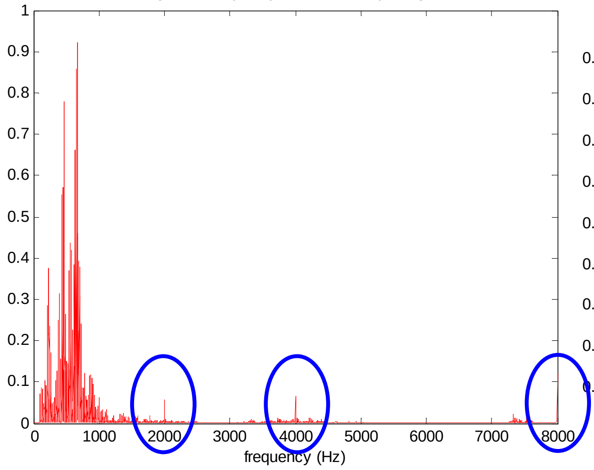


Fig 2.6 Frequency content of output signal

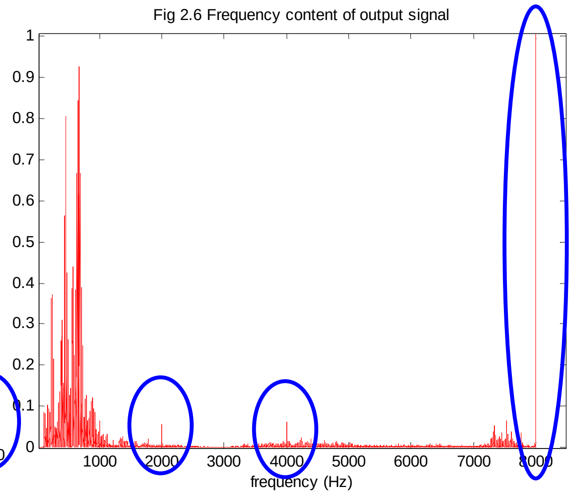


Fig 2.8 Frequency content of output signal

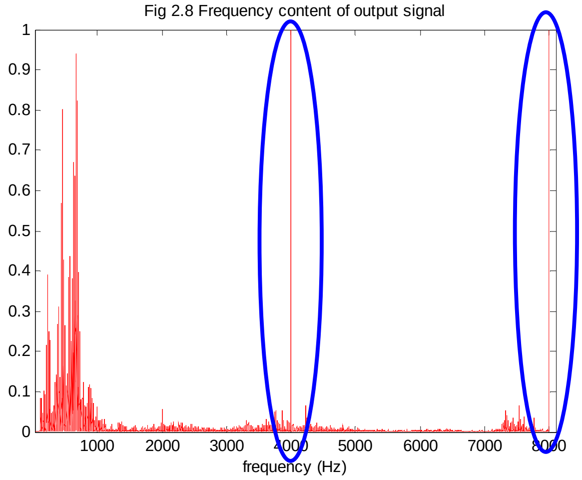
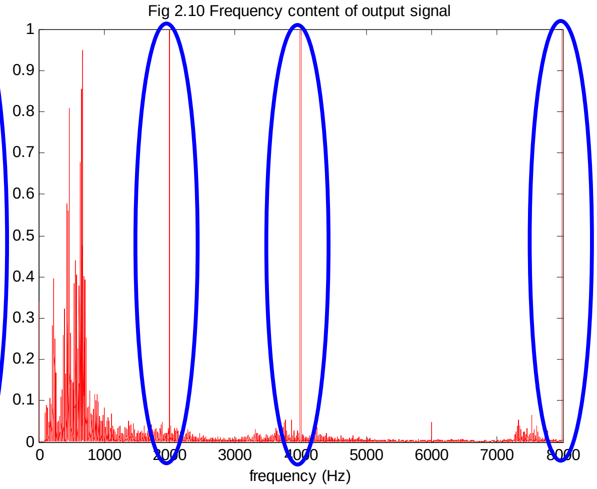


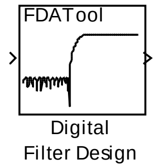
Fig 2.10 Frequency content of output signal

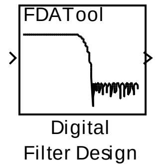


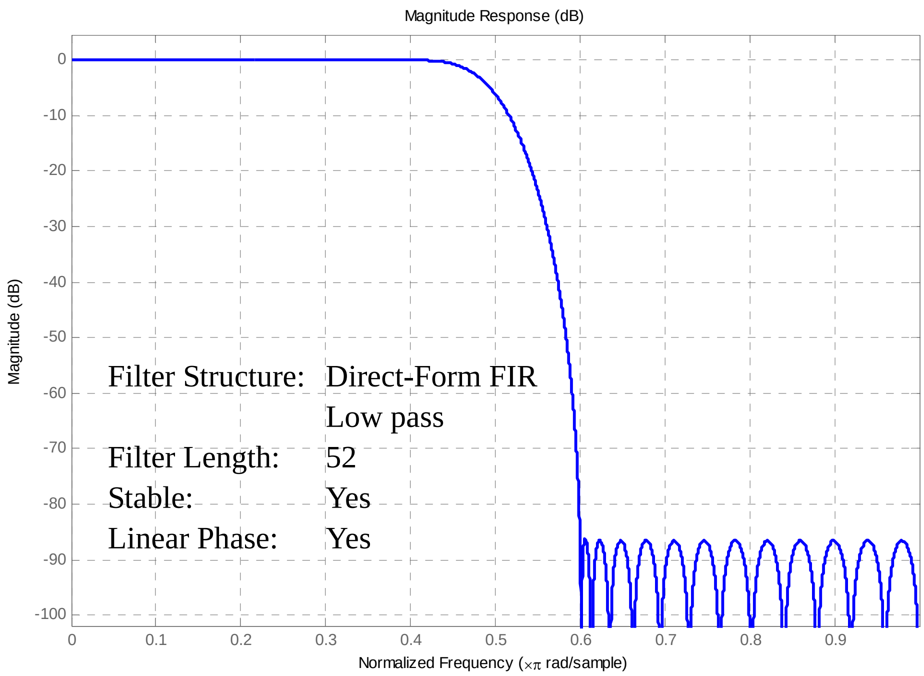
The blue circles indicate that there exist noises in certain frequencies among the output signal. Moreover, the noises frequencies are exactly the boundaries between different sub-bands. And these noises get bigger and bigger if the number of bits is getting smaller and smaller. In Fig 2.10, there are huge noises in 2000 Hz, 4000 Hz and 8000 Hz. The magnitudes of these noises are much larger than the signal itself. As a result, the SNR drop to -21.8679 dB.

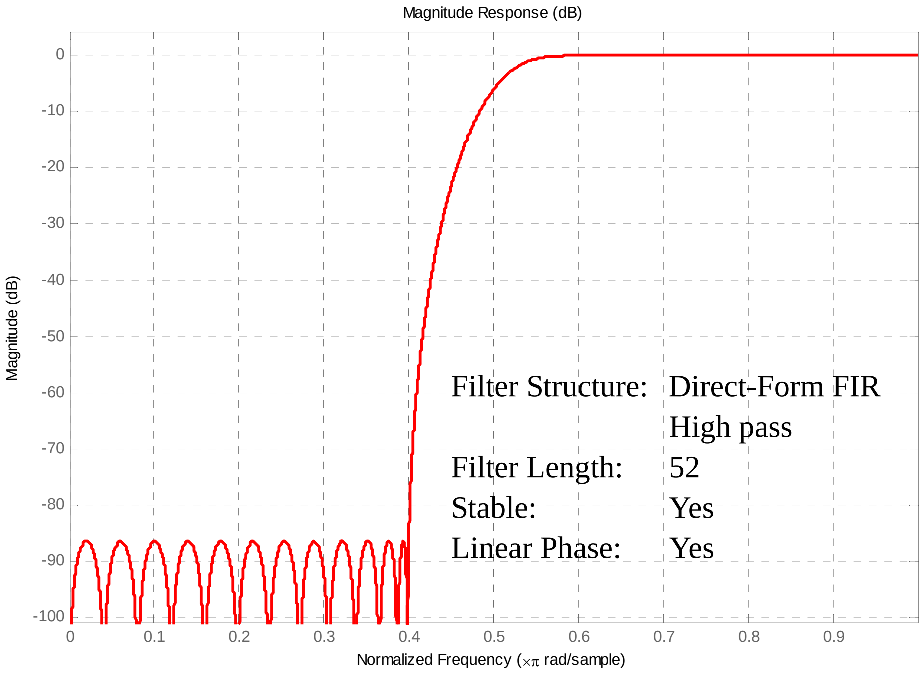
2.2 Simulation version #2

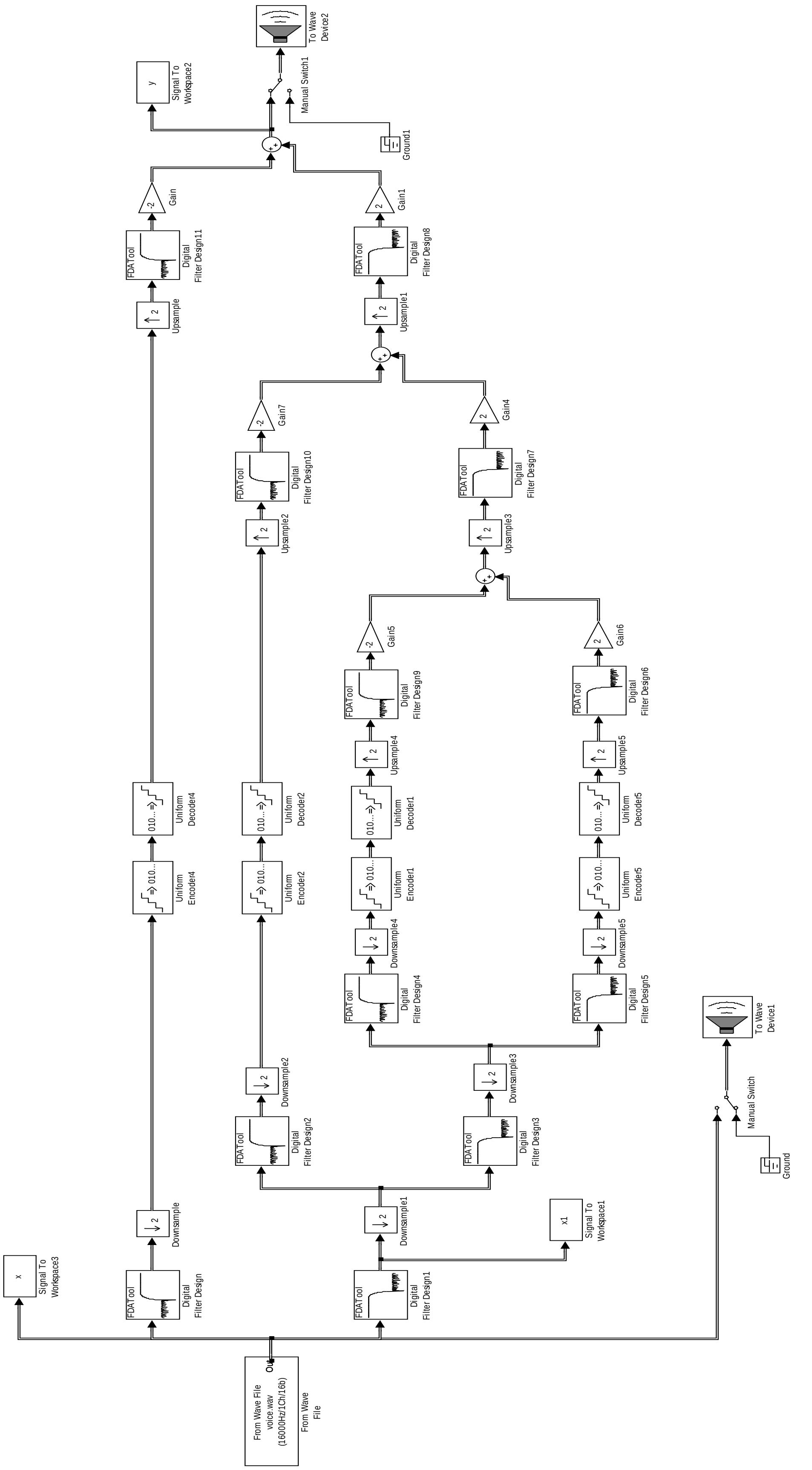
In this version, I use the same structure as version #1, however, instead of Two-Channel Analysis/Synthesis Sub-band Filter, I designed the filters using FDA tool of Matlab. The lowpass filter is a 52nd FIR filter and the highpass filter is also a 52nd FIR filter. To satisfy the perfect reconstruction and alias-free conditions, a “-2” and “2” gains were added at the end of each sub-band. Also, since the up-sample and down-sample parts were not included, I use the corresponding blocks in Simulink.








Magnitude Response (dB)
Magnitude (dB)
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
-100
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
Normalized Frequency (×π rad/sample)
Filter Structure: Direct-Form FIR
High pass
Filter Length: 52
Stable: Yes
Linear Phase: Yes


From Wave File
voice.wav
(16000Hz/1Ch/16b)
Out
From Wave File
x
Signal To Workspace3
FDATool
Digital Filter Design
Downsample
Uniform Encoder4
Uniform Decoder4
Upsample
Digital Filter Design11
Gain
y
Signal To Workspace2
Manual Switch1
To Wave Device2
Ground1
Gain1
Digital Filter Design8
Upsample1
Gain7
Digital Filter Design10
Upsample2
Uniform Decoder2
Uniform Encoder2
Downsample2
Digital Filter Design2
Digital Filter Design1
Downsample1
x1
Signal To Workspace1
Digital Filter Design3
Downsample3
Digital Filter Design4
Downsample4
Uniform Encoder1
Uniform Decoder1
Upsample4
Digital Filter Design9
Gain5
Digital Filter Design5
Downsample5
Uniform Encoder5
Uniform Decoder5
Upsample5
Digital Filter Design6
Gain6
Upsample3
Digital Filter Design7
Gain4
To Wave Device1
Manual Switch
Ground

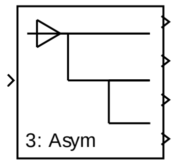
SNR vs. Number of bits in encoder/decoder

Sub-band encoder/decoder	SNR	Frequency content of input signal	Frequency content of output signal
16(L)+16+16+16(H)	21.3904 dB	Fig 2.4	
16(L)+16+16+8(H)	19.8382 dB		
16(L)+16+8+8(H)	19.7255 dB		
16(L)+8+8+8(H)	19.6167 dB		
8(L)+8+8+8(H)	18.8598 dB		
8(L)+8+8+4(H)	-22.8241 dB		
8(L)+8+4+4(H)	-23.2859 dB		
8(L)+4+4+4(H)	-23.6378 dB		
4(L)+4+4+4(H)	-25.4654 dB		

The reason why the frequency content of output signal was not shown is that they are almost the same as in Simulation version #2. There exist noises in certain frequencies among the output signal. Moreover, the noises frequencies are exactly the boundaries between different sub-bands. And these noises get bigger and bigger if the number of bits is getting smaller and smaller.

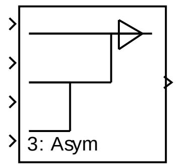
2.3 Simulation version #3

In this version, a Dyadic Analysis Filter Bank and a Dyadic Synthesis Filter Bank were used.



Dyadic Analysis Filter Bank

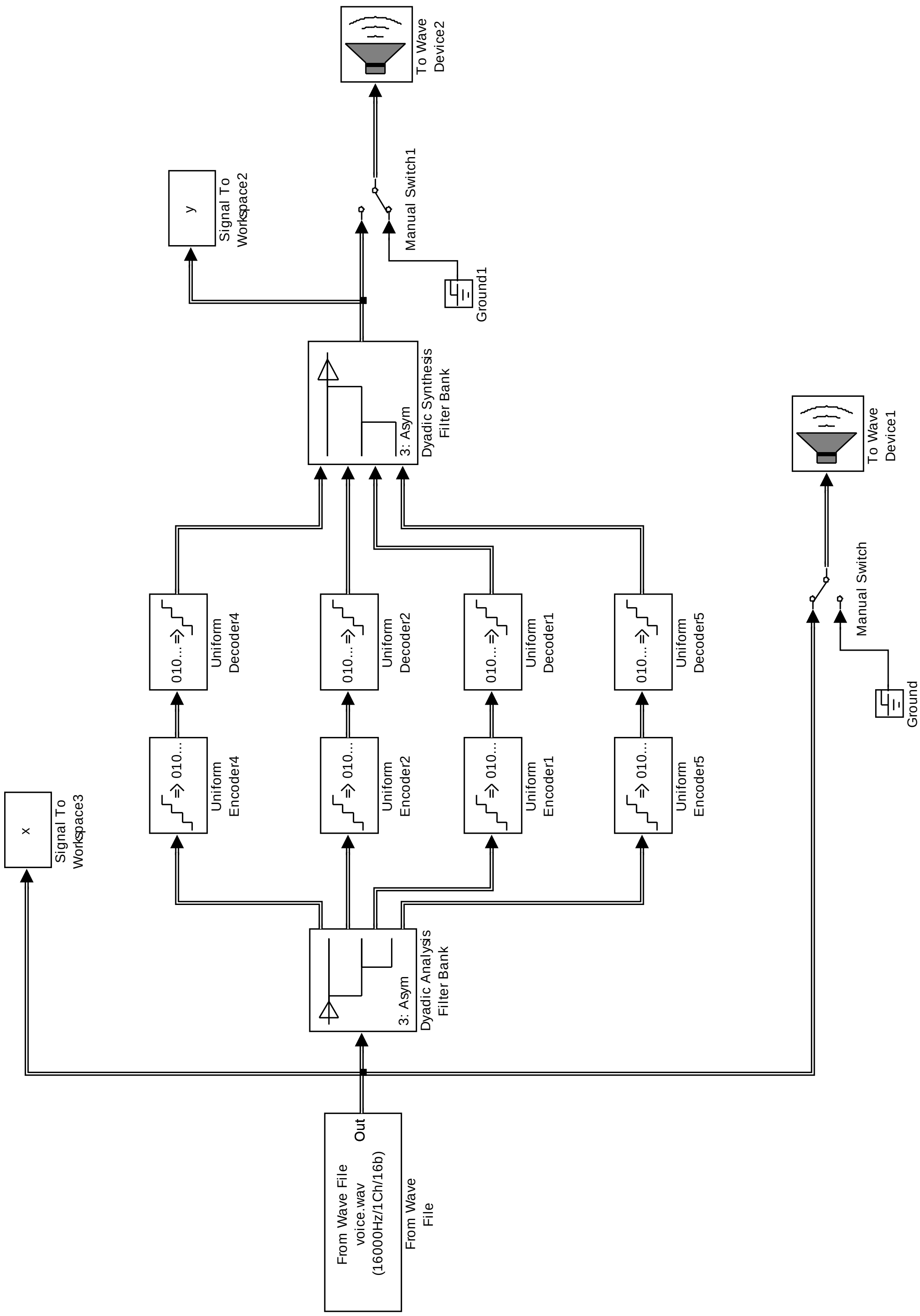
The Dyadic Analysis Filter Bank decomposes a signal into subbands with smaller bandwidths and slower sample rates. Uses a filter band with specified lowpass and high pass FIR filters, which can be user-defined or wavelet-based. The lowpass and highpass filters are half-band filters designed to complement each other.



Dyadic Synthesis Filter Bank

The Dyadic Synthesis Filter Bank reconstructs a signal from its subbands with smaller bandwidths and slower sample rates.

I choose Haar and Biorthogonal filter respectively. The following tables show the result of SNR vs. Number of bits in encoder/decoder


From Wave File
voice.wav
(16000Hz/1Ch/16b)
Out
From Wave File
Signal To Workspace3
x
3: Asym
Dyadic Analysis Filter Bank
Uniform Encoder4
Uniform Encoder2
Uniform Encoder1
Uniform Encoder5
Uniform Decoder4
Uniform Decoder2
Uniform Decoder1
Uniform Decoder5
3: Asym
Dyadic Synthesis Filter Bank
y
Signal To Workspace2
Manual Switch1
Ground1
To Wave Device2
Manual Switch
Ground
To Wave Device1

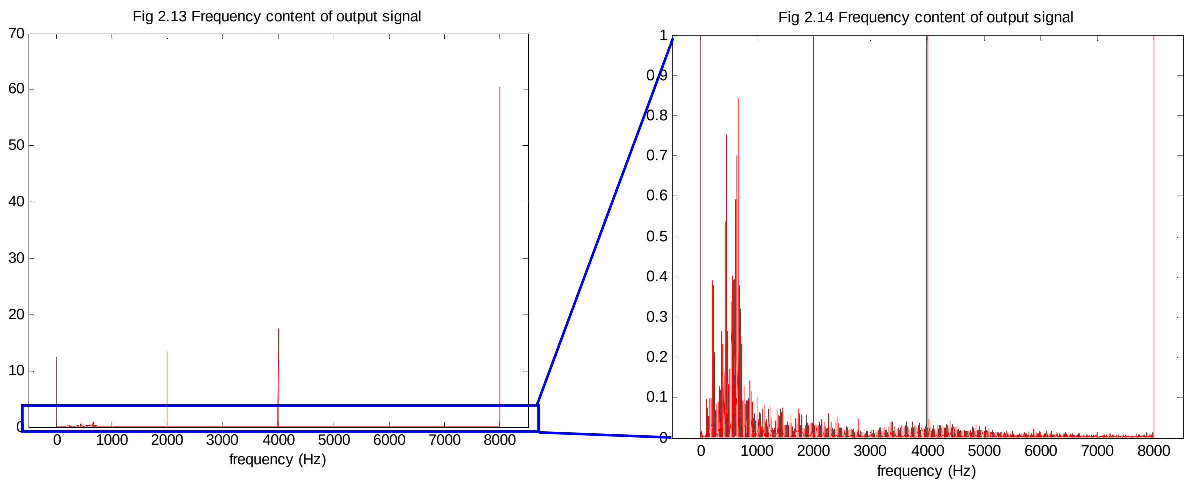
Filter: Haar

Sub-band coder/decoder	SNR	Frequency content of input signal	Frequency content of output signal
16(L)+16+16+16(H)	15.1394 dB	Fig 2.4	
16(L)+16+16+8(H)	15.0544 dB		
16(L)+16+8+8(H)	15.0275 dB		
16(L)+8+8+8(H)	15.0208 dB		
8(L)+8+8+8(H)	14.9874 dB		
8(L)+8+8+4(H)	-16.5630 dB		
8(L)+8+4+4(H)	-17.1211 dB		
8(L)+4+4+4(H)	-17.2220 dB		
4(L)+4+4+4(H)	-17.4179 dB		

Filter: Biorthogonal

Sub-band coder/decoder	SNR	Frequency content of input signal	Frequency content of output signal
16(L)+16+16+16(H)	23.8850 dB	Fig 2.4	
16(L)+16+16+8(H)	23.1971 dB		
16(L)+16+8+8(H)	23.0936 dB		
16(L)+8+8+8(H)	23.0183 dB		
8(L)+8+8+8(H)	22.8717 dB		
8(L)+8+8+4(H)	-16.9292 dB		
8(L)+8+4+4(H)	-17.5945 dB		
8(L)+4+4+4(H)	-17.9540 dB		
4(L)+4+4+4(H)	-18.1012 dB		Fig 2.13 & 2.14

The same thing happened when I used fewer bits in quantization.



Four major noises are in 0 Hz, 2000 Hz, 4000 Hz and 8000 Hz. They have considerable magnitudes which are much bigger than the signal itself. Based on this simulation, the total number of bits should not be smaller than 32-bit; otherwise the SNR will be negative. However, the input signal is a 16-bit, 16000 Hz human voice. I'm not sure why this happened.

Appendix A

```
% -----------------------------------------------------------------------------------------------
% Calculate the SNR
% -----------------------------------------------------------------------------------------------
% -----------------------------------------------------------------------------------------------
% Clear workspace & command window
% -----------------------------------------------------------------------------------------------
clc;
% -----------------------------------------------------------------------------------------------
% FFT of input signal
% -----------------------------------------------------------------------------------------------
figure(1);
X = fft(x, 65536);
Pxx = X.* conj(X) / 65536;
f = 16000 * (0:32768)/65536;
plot(f, Pxx(1:32769)); hold on;
title('Frequency content of input signal');
xlabel('frequency (Hz)');
% -----------------------------------------------------------------------------------------------
% FFT of output signal
% -----------------------------------------------------------------------------------------------
figure(2);
Y = fft(y, 65536);
Pyy = Y.* conj(Y) / 65536;
f = 16000 * (0:32768)/65536;
plot(f, Pyy(1:32769),'r'); hold on;
title('Frequency content of output signal');
xlabel('frequency (Hz)');
% -----------------------------------------------------------------------------------------------
```

% SNR calculation based on $SNR = 10 \cdot \log_{10}(Energy(x)/Energy(y-x))$

```
% -----------------------------------------------------------------------------------------------
sum1 = 0;
sum2 = 0;
for i = 1:numel(Pxx)
    sum1 = sum1 + Pxx(i)^2;
    sum2 = sum2 + ((Pyy(i)) - (Pxx(i)))^2;
end
SNR = 10 * log10((sum1) / (sum2));
disp(SNR);
```