Digital Signal Processing: Experiment 4

Study and Analysis of DTMF coder/decoder using Digital FIR Filter

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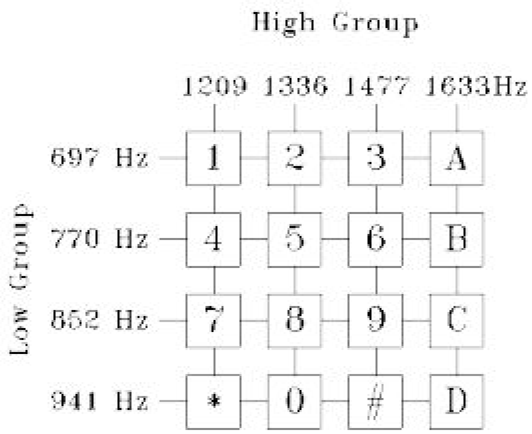
Aim of the experiment:

- Study the functionality of DTMF and its different applications.
- Implement bandpass digital filters
- Implement DTMF coder/decoder
- Characterise a filter based on its response to different frequency components in the input

Theory:

Dual Tone Multi Frequency is the basis for telephone system. Each key is mapped to a specified column and row frequency. As the key is pressed, two sinusoids (tones) are generated. These two tones are summed and this “Dual tone” signal now represents the pressed key. The two tones are carefully chosen so that no harmonics occur ie. No frequency is an integral multiple of the other and the sum/ difference of these two frequencies does not equal to any of the frequencies.

Eg: Pressing the digit 2 will generate the tones 1336 Hz and 697 Hz.



Decoding: The dual tone signal is now passed through a filter-bank containing 8 filters. Each filter is a band pass filter with the central frequency tuned to a tone and appropriately sharp cut-off to prevent overlapping with the adjacent tones. As the signal passes through the filter bank, the outputs of the filters tuned to the tones of the input signal are high in amplitude, the rest of the filter outputs are low amplitude outputs. With a simple thresholding scheme, the outputs are now 1s and 0s (digital output). Now this can be decoded using appropriate digital logic to get back the numbers.

For successful decoding, since we do not know the strength of the signal and the output comes over a channel and is hence susceptible to noise, we need the following criteria:

1. $(P_a + P_b) > 0.75*P_t$

2. $\Delta f/f < 1.5\%$

The frequency content in the output is estimated using the Goertzel algorithm, which fills up frequency bins that are predetermined by the user with the frequency content. Taking the max. gives us the frequency content in one such bandpass filter output (since only one tone could possibly result as a filter output).

Goertzel algorithm proves computationally more efficient than FFT for small number of data points. For our consideration, each symbol lasts for 40ms and we have 8000*0.04=320 data points, which is quite less. Further, it has been shown that to minimise the error, this may be chosen as low as N=205, rejecting the other data points.

Goertzel algorithm is most typically used as a hardware implemented replacement for the filter bank, as designing such sharp bandpass filters isn't always feasible. This block can directly find the two peaks and estimate the frequency content of the signal, making it an alternative DTMF decoder.

Code Snippets:

These snippets are pseudo codes.

```
//a if else loop has been used to detect the pressed key
if(k=='1')
    x=sin(2*pi*697*t)+sin(2*pi*1209*t);
elseif(k=='2')
    x=sin(2*pi*697*t)+sin(2*pi*1336*t);
// And so on…


//filter definition
L=65;%order of filter
w=-pi:.001:pi;
n=0:1:L;
% 8 filters
h=zeros(8,L+1);
%filter functions
for i=1:8
    h(i,:)=cos(wc(i)*n);
end
%array for storing maximum values of filter ouputs
S=zeros(1,8);

S(j)=max(abs(fft(y(j,:))));%storing the maximum value of a certain filter output
end
symbol = {'1','2','3','A';'4','5','6','B';'7','8','9','C';'*','0','#','D'};
% determining frequency components present in each segment
a=find(S==max(S)); % finding 1st maximum
S(a)=0;
b=find(S==max(S)); %finding 2nd maximum
if(a>b)
    a=a+b;
    b=a-b;
```

```
    a=a-b;
end
//based on values of a and b, a decoding logic is implemeted.
```

Observations and Plots:

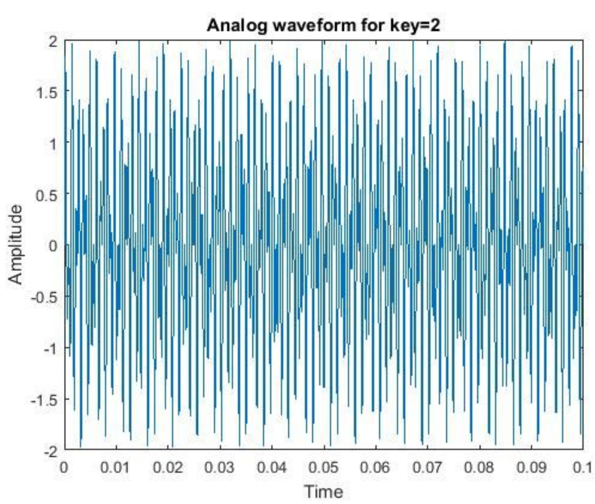
We have plotted the waveform sampled at 10000 Hertz for each key pressed

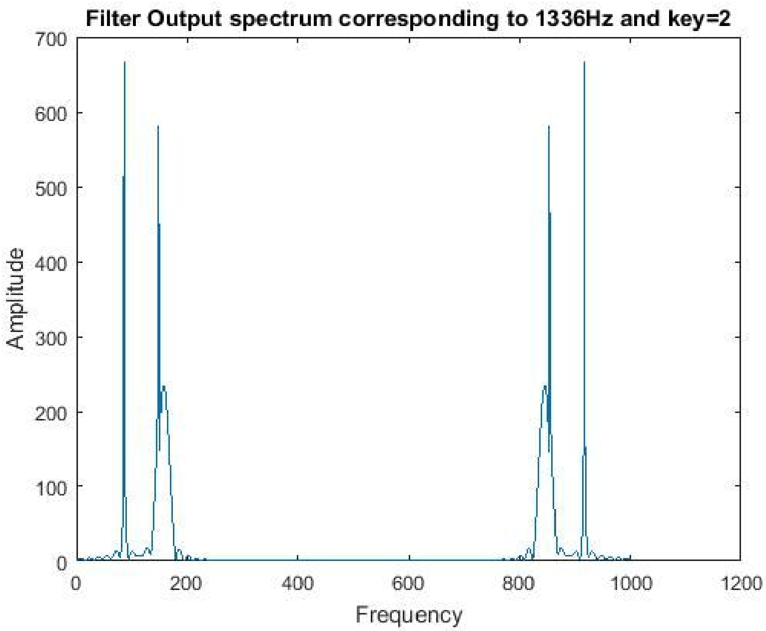
x(t) = sin(2 * pi * f1 * t) + 6cos(2 * pi * f2 * t)

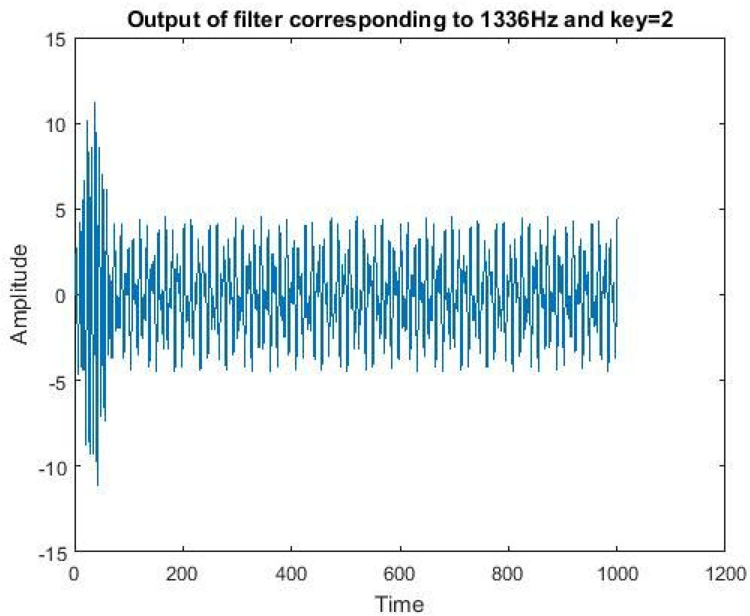
FIR filter bank consists of 8 Band Pass Filters, each of which passes only one frequency components. The input signal for all filters is the same DTMF signal. From the ouput signals of filters, frequency components present in the input signal are determined.

Output at the terminal:

```
>> Experiment_4
Press any key4
frequency components present are
770
1209
number typed is 4
```




Filter Output spectrum corresponding to 1336Hz and key=2
Amplitude
Frequency
0
100
200
300
400
500
600
700
0
200
400
600
800
1000
1200


Output of filter corresponding to 1336Hz and key=2
Amplitude
Time
15
10
5
0
-5
-10
-15
0
200
400
600
800
1000
1200

The DTMF generated is with respect to random sequence of random length. To account for input noise, we have also added AWGN in the input signal (Additive White Gaussian Noise). This corrupts the input signal and pushes up the level of Power Spectral Density equally everywhere. As a result, the power criterion may fail as if $P_a + P_b > 0.75*P_t$, it is not necessary that

$$(P_a + P_b + 2P) > 0.75*(P_t + 8P)$$

Here, we do not have a possibility of frequency deviation, as AWGN does not corrupt relative frequency content, but pushes the level at each frequency equally. However, if we modelled channel non-linearity and distortion, we would also get some frequency deviation and the frequency criterion could also be violated.

Discussions:

- Due to inherent non-linearity of the channel, we have intermodulation distortion, which causes second and higher order terms in the spectrum of output. So, for successful decoding, we have to ensure that the two input tones are such that the output when band pass filtered does not give an erroneous decoded value. For this we ensure that the second and higher order terms do not coincide with any of the other frequencies. So, we choose f's such that no frequency is an integral multiple of the other and the difference or sum of any two frequencies should not equal any of the frequencies.

- Here the sampling frequency is chosen as 10kHz, and as 10/2 > max(f)= 1633, it satisfies the Nyquist criterion and the band pass filtered output will be as desired. In practice, since the channel on which the dual tone is transmitted is the same as the audio channel (telephone line routed through operator), the sampling frequency will be the same as the standard for audio.

- Depending upon the proximity of the frequencies, the required sharpness of bpf and threshold applied for decoding will change. If they are closer, we'll need higher order bpf and a higher thresholding for calculating binary.

- Depending on the selectivity of the bandpass filter, we'll have different power distributions. So, to meet the power criterion, it is imperative that we use a high order filter (95 in our case). However, if we have a frequency deviation, we would need to estimate this as we want to see if frequency criterion is satisfied. For this, we cannot have a very sharp filter. This high order and contradiction is further motivation for us to ditch the filter bank and go for the Goertzel algorithm implementation.

- We have taken the sequence length varying from 5 to 30. If we took a larger length, we would have lesser probability of correct detection of the sequence for a given SNR.