Experiment 15
FFT for Frequency Detection

Aim

To find the frequency contents of the speech signal using FFT.

Theory

Analog signal $x(t)$ when sampled using sampling frequency F results in a sampled signal $x(n)$. We are assuming that the sampling frequency is above the Nyquist rate so that there is no aliasing. When a speech signal is interfaced to our computer system via a sampler and analog-to-digital converter, we will obtain a digital signal. We will take FFT of the speech data segment and will try to plot the amplitude with respect to frequency graph to identify the frequency contents of the speech segment.

Experiment

We will first record a speech signal in our own voice using a sound recorder facility of the computer system. The speech signal will be recorded in the ".wav" file format. Here, it has a 44 byte header containing the information regarding the sampling frequency, data format (if 8-bit or 16-bit, if mono or stereo) and number of data values. In case we want to process the speech file, we will have to first remove the header and track the start of the utterance. When we record a speech file, there is a silence part at the start, because we take some time to utter the word after record command is given.

Let us take first 1024 samples of the speech data file and take its 1024-point FFT. According to the symmetry property of FFT, the FFT sample will exhibit symmetry about its center value, that is, 512th sample. We will use this symmetry in the FFT output and will concentrate only on the first 512 samples of FFT output.

Let us calibrate the frequency axis. 512 samples of FFT output correspond to the Nyquist frequency that is $F/2$, where F is the sampling frequency. The nth sample of FFT will correspond to $(F/2) \times (n/512)$. Let us now plot FFT output with respect to this frequency so that we can directly read out the frequency values from FFT plot ($F = 16{,}000$ Hz).

The reader is first supposed to write a MATLAB program to open the speech file using "fopen" command and then read a file using "fread" command and plot it. We can track the start of the utterance manually. Using "fseek" command, track the start of the utterance. The reader can then take 1024 data samples and take FFT, calibrate frequency axis and plot it.

The speech signal plot for 1024 samples is shown in Figure 1. The FFT plot calibrated for frequency in hertz is shown in Figure 2. We see from Figure 2 that only frequency contents up to 2000 Hz are present in speech signal.

The MATLAB program is as follows.

```
%experiment no 15.
%to find frequency contents of speech signal
clear all;
fp=fopen('watermark.wav','r');
fseek(fp,44,-1);
a=fread(fp,128);
plot(a);title('plot of speech signal');xlabel('sample number');
ylabel('Amplitude');
b=fft(a);
figure;
c=abs(b);
for i=1:127,
    d(i)=c(i+1);
f(i)=(i/128)*8000;
end
plot(f,d);title('plot of frequency contents of speech
signal with resolution 62.5 Hz');xlabel('frequency in Hz');
ylabel('Amplitude');
```

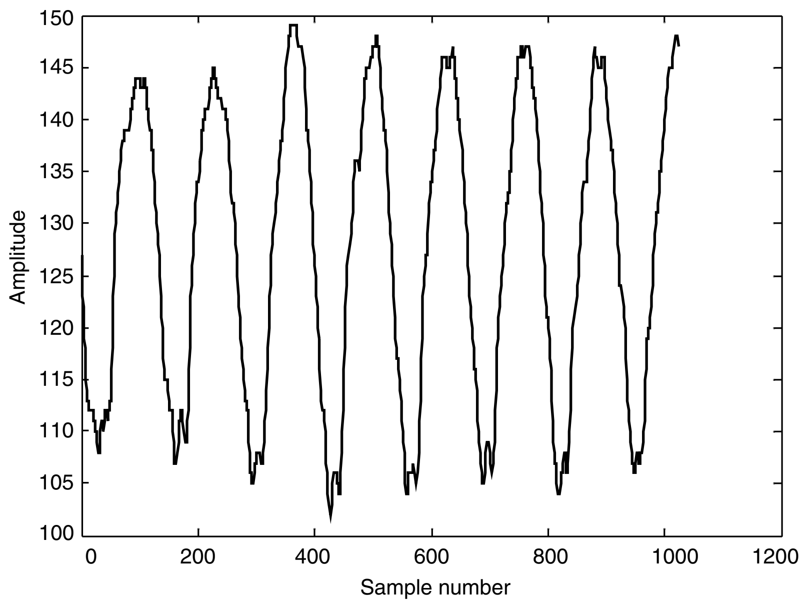

Figure 1 Plot of 1024 samples of speech signal sampled with $F = 16$ kHz.

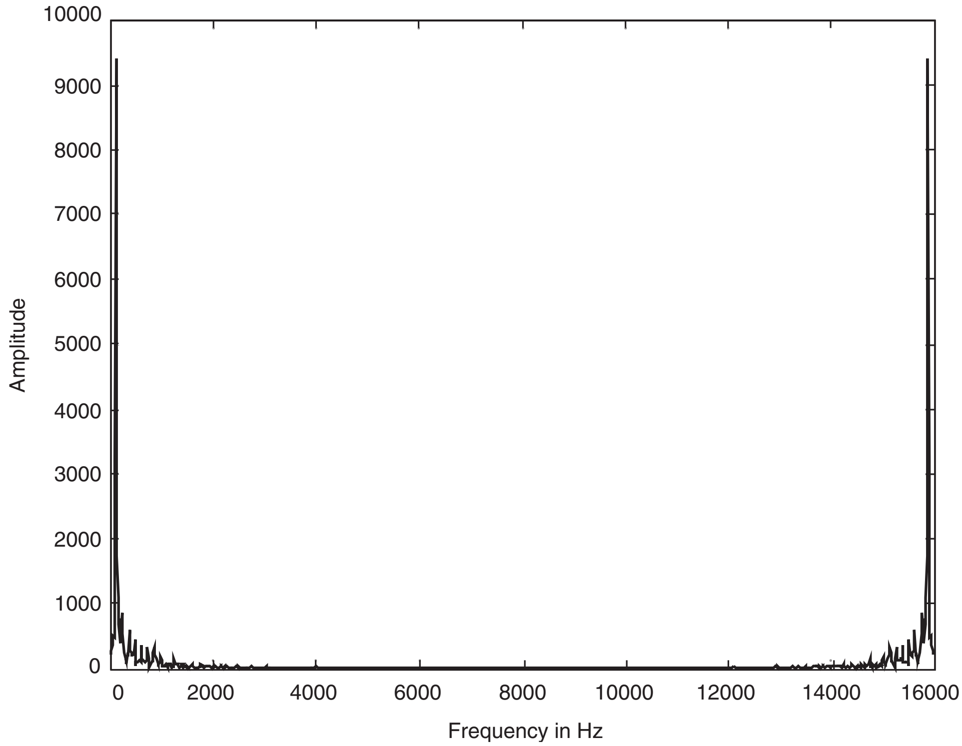

Figure 2 Plot of amplitude with respect to frequency graph. Frequency is in Hertz.