# Experiment 16
# Filtering Speech Using FIR LPF

## Aim

To apply FIR LPF to filter the signal.

## Theory

We will consider a speech signal same as in Experiment 15. Let us consider FIR filter designed such as in Experiment 8 and filter the speech signal.

## Experiment

Let us consider 1024 samples of speech data and convolve it with 11 filter coefficients.

We have used following specifications for the FIR filter.

1. Sampling frequency: 16 kHz.
2. Cut-off frequency: 2 kHz.
3. Duration of the response: 0.00125 s.

**Teaser** *The reader has to write a program in MATLAB to convolve the two sequences and plot the output. Note that we will get 11 FIR filter coefficients and these will be convolved with the signal. The reader is encouraged to verify the filter design.*

Figure 1 shows a plot of original speech signal and Figure 2 shows the plot of filtered speech signal. We can see that in Figure 2 high frequency contents of the signal are removed. The MATLAB program is as follows.

```
%experiment no 16.
%filtering of speech signal using FIR-LPF
clear all;
fp=fopen('watermark.wav','r');
fseek(fp,44,-1);
a=fread(fp,1024);
plot(a);title('plot of speech signal');xlabel('sample number');
ylabel('Amplitude');
fs=16000;
```

```
t=0.00125;
Q=fs/2*t;
disp(Q);
for i=1:10,
    x(i)=(sin(0.125*pi*i))/(i*pi);
end
a(11)=0.125;
for i=1:10,
    a(i)=x(Q-(i-1));
end
for i=2:11,
a(i+Q)=x(i-1);
end
w1=window(@blackman,21);
for i=1:21,
    z(i)=a(i)*w1(i);
end
c=conv(a,z);
figure;
plot(c);axis([1 1024 80 130]);title('plot of filtered speech
signal');xlabel('sample number'); ylabel('Amplitude');
```

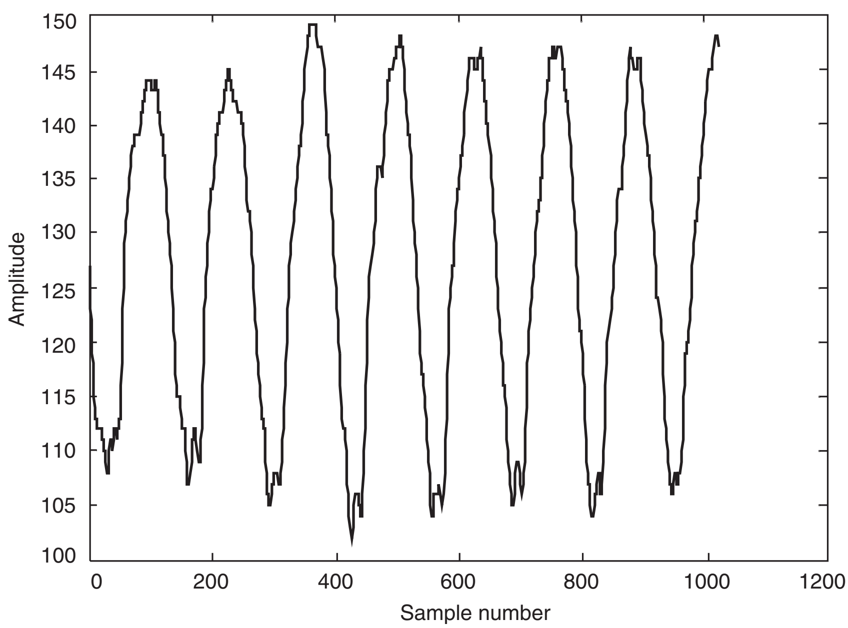


**Figure 1** Plot of original speech signal.

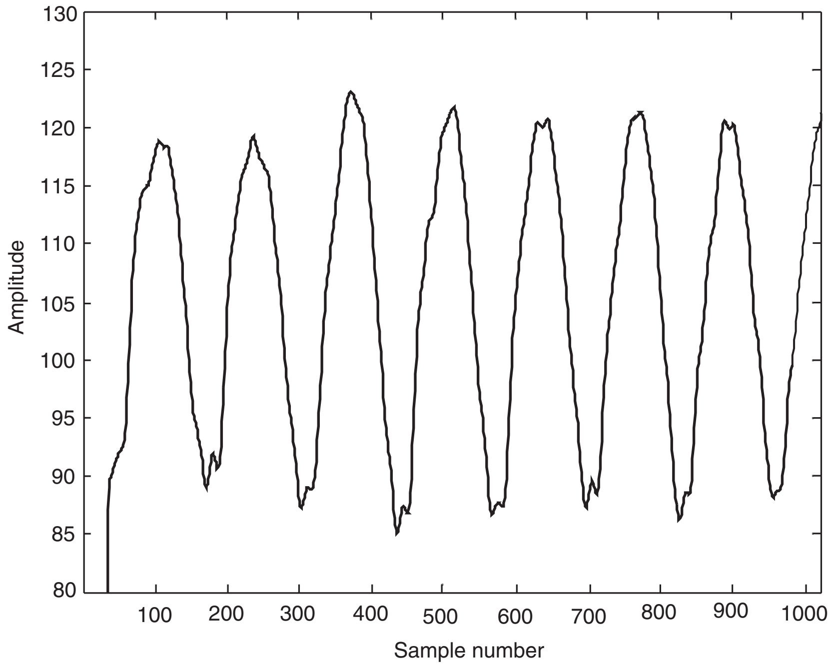


**Figure 2** Plot of filtered speech signal.

***Note:*** The signal is seen to be smooth.