MATLAB Programs

Program 1

Write a program to filter speech signal using FIR-LPF and decimating it by factor of 2.

```
%filtering of speech signal using FIR-LPF and decimating it
by factor of 2
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');
% we have plotted 1024 samples of speech
fs=8000;
Q=10;
for i=1:10,
    x(i)=(sin(0.5*pi*i))/(i*pi);
end
a(11)=0.5;
for i=1:10,
    a(i)=x(Q-(i-1));
end
for i=2:11,
a(i+Q)=x(i-1);
end
%  we  have  designed  the  filter  coefficients  using  FS
expansion method
% cut off frequency is 2 KHZ
w1 = window(@blackman,21);
for i=1:21,
    z(i)=a(i)*w1(i);
end
% we have used Blackman windowing
c=conv(a,z);
%we have filtered the speech signal
figure;
plot(c); title('plot of filtered speech signal');xlabel('sample
number'); ylabel('Amplitude');
%we are plotting filtered speech output
for i=2:2:1024,
    d(i/2)=c(i);
end
%we have down sampled the output
figure;
```

```
plot(d); title('plot of decimated speech signal');xlabel('sample
number'); ylabel('Amplitude');
% we have plotted output of a decimator
```

Output

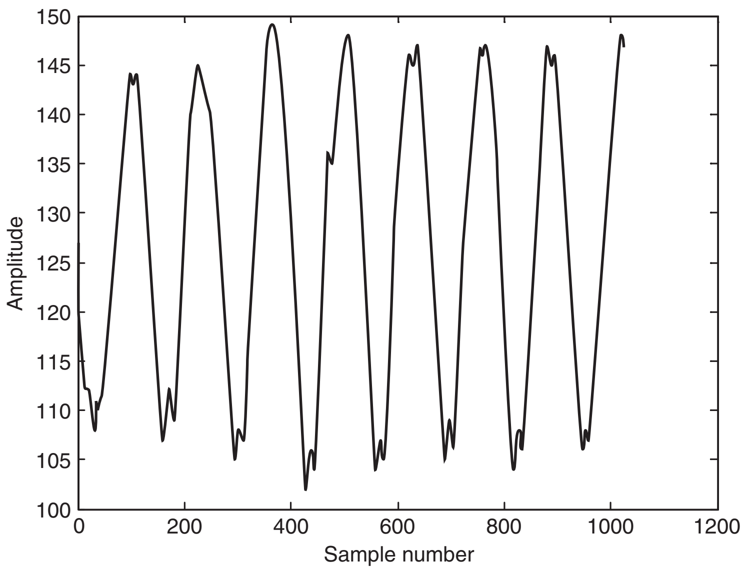

Figure 1 Plot of speech signal.

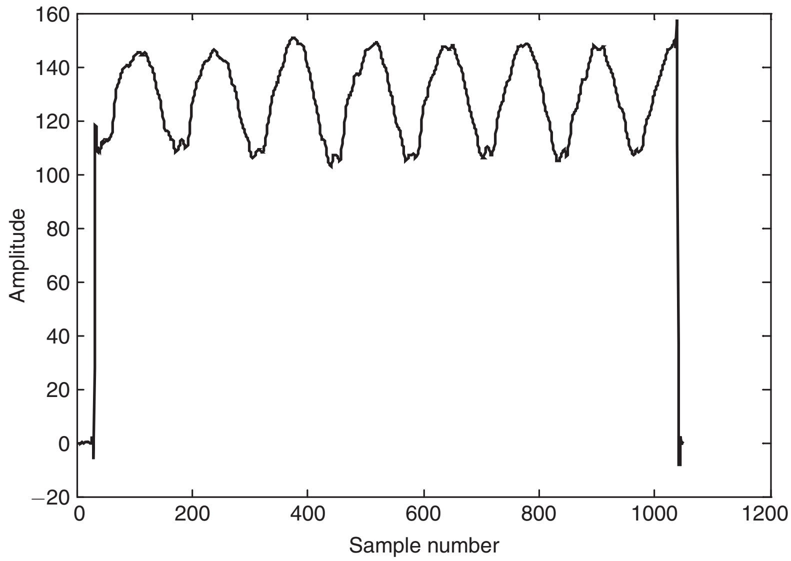

Figure 2 Plot of the filtered speech signal with cut-off frequency of 2 kHz.

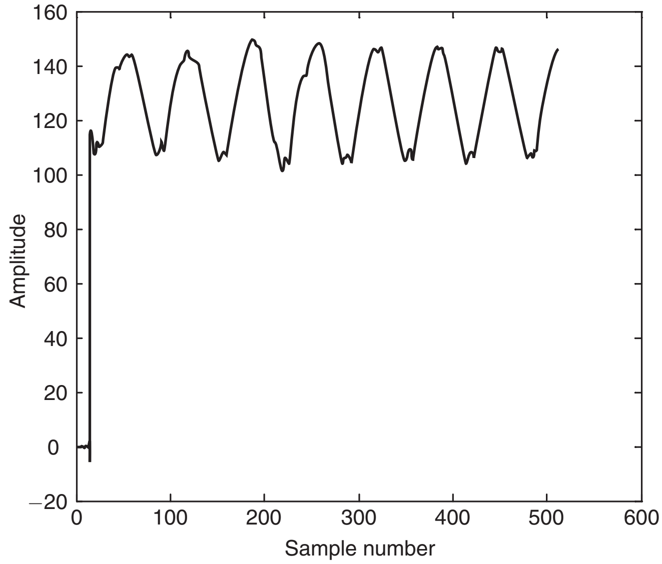

Figure 3 Plot of decimated speech signal now containing only 512 samples.

Program 2

Write a program to interpolate the speech signal by a factor of 2 and filter it.

```
%Interpolating the speech signal by a factor of 2 and
filtering it
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');
% we have plotted 1024 samples of speech
 for i=1:1024,
     d(2*i)=a(i);
 end
for i=1:2:2048,
    d(i)=0;
end
figure;
plot(d);axis([1 2048 100 160]);title('plot of up sampled
speech signal');xlabel('sample number');
ylabel('Amplitude');
%we have up sampled the output and plotted it
fs=8000;
```

```
Q=10;
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
%  we  have  designed  the  filter  coefficients  using  FS
expansion method
% cut off frequency is 2 KHZ
w1 = window(@blackman,21);
for i=1:21,
    z(i)=a(i)*w1(i);
end
% we have used Blackman windowing
c=conv(d,z);
%we have filtered the speech signal
figure;
plot(c);title('plot of filtered speech signal');xlabel('sample
number'); ylabel('Amplitude');
%we are plotting filtered speech output
```

Output

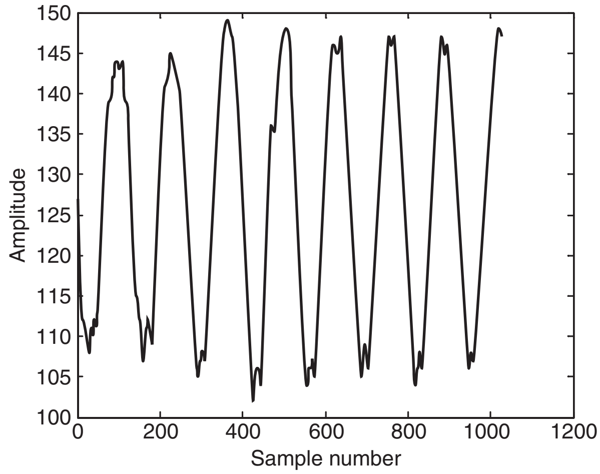

Figure 4 A plot of original speech signal.

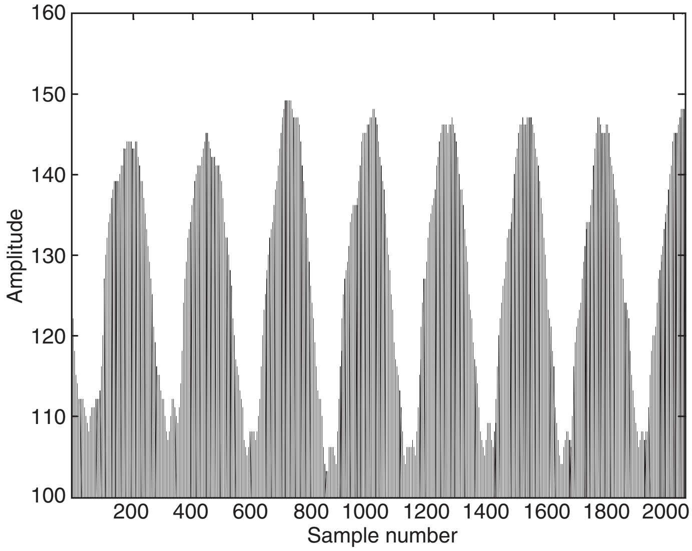

Figure 5 A plot of upsampled signal.

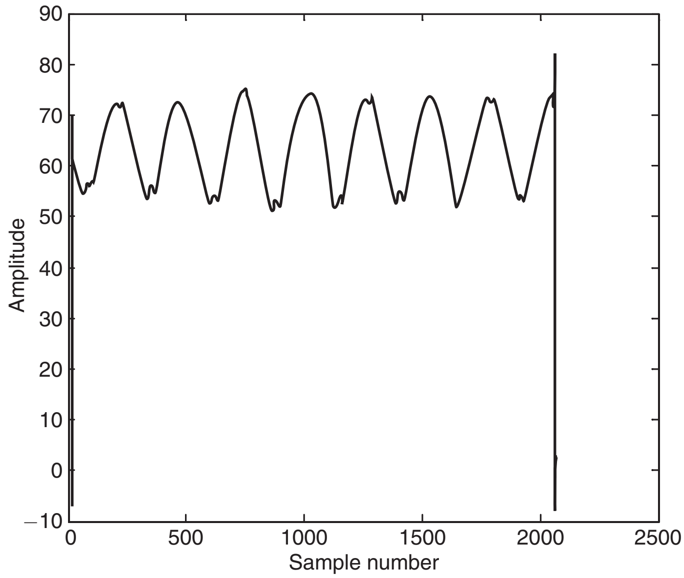

Figure 6 A plot of filtered speech signal as a result of interpolation.