Introduction to Signal Analysis

(1) mylab1_sig1.m

```
clear all;
close all;
clc;

L1 = 60; %signal length
fs1 = 20; %sampling frequency for discrete time signal
f1 = 3; %signal frequency
t1 = [0:L1-1]/fs1; %signal sampling instants
x1 = cos(2*pi*f1*t1); %signal
stem(t1,x1); %plot
xlabel('Time (s)');ylabel('Amplitude');
title('Cosine signal (3Hz) sampled at 20Hz');
```

#Exercise:

(a)Sampling period of x1 = (1/fs1) = 1/20 =0.05

%(b)

```
    f = [4 5 7 16 15 13 17];
    [p,q] = size(f);
    figure
    for i=1:q
        ti = [0:L1-1]/fs1;
        xi = cos(2*pi*f(i)*ti);
        subplot(3,3,i);stem(ti,xi);
        xlabel('Time (s)');ylabel('Amplitude');
        title('Cosine signal sampled at 20Hz');
    end
```

%(c) Signals with frequency greater than 10 Hz is not obeying the Nyquist
% Criterion (i.e. 16Hz,15Hz and 17Hz)

%(d) The max signal frequency allowed in order to avoid aliasing
% f = fs/2 = 20/2 = 10Hz
% If we exceed the max allowed frequency then high freq component will
% be aliased(overlapping of spectrum occurs) with low freq term and high freq terms get lost.

(2) mylab1_sig2.m

```
clear all;
close all;
clc;

L1 = 60; %signal length
fs1 = 20; %sampling frequency for discrete time signal
f1 = 3; %signal frequency
t1 = [0:L1-1]/fs1; %signal sampling instants

x1 = cos(2*pi*f1*t1); %signal
x2 = cos(2*pi*f1*t1 + pi/2);% pi/2 phase shifted signal
subplot(221);stem(t1,x1);xlabel('Time (s)');ylabel('Amplitude');
title('Signal x1');
subplot(222);stem(t1,x2);xlabel('Time (s)');ylabel('Amplitude');
title('Signal which is pi/2 phase shifted with x1');
```

```
x12 = x1 + x2;        % combination of x1 and x2
subplot(223);stem(t1,x12);
xlabel('Time (s)');ylabel('Amplitude');title('Combination of x1 and x2');

x3 = cos(2*pi*f1*t1 + pi); % pi phase shifted signal
x13 = x1 + x3;     % combination of x1 and x3
subplot(224);stem(t1,x13);xlabel('Time (s)');ylabel('Amplitude');
title('Combination of x1 and x3(pi phase shifted with x1)');
```

#Exercise:

% When x1 and x3 is combined then:

% x1 + x3 = x1 + (-x1) = 0

% Theoretically it should be zero and the result obtained above justifies

% it very closely.

(3) Signals_DSP.m

```
clear all;
clear all;
clc;

% Signal xa - (Rectangular signal)
La = 20; %signal length
na = [0:La-1]; %signal sample index
xa = [zeros(1,La/4) ones(1,La/2) zeros(1,La/4)];  %Rectangular signal
subplot(221);stem(na,xa);
xlabel('n');ylabel('Amplitude');
title('Rectangular signal');

% Signal xb - (Unit Impulse signal)
xb = [ones(1,1) zeros(1,La-1)]; % Unit Impulse signal
subplot(222);stem(na,xb);
xlabel('n');ylabel('Amplitude');
title('Unit Impulse signal');

% Signal xc - (Sinc signal)
nb = [-La/2:La/2-1];
xc = sinc(0.2*pi*nb);
subplot(223);stem(nb,xc);
xlabel('n');ylabel('Amplitude');
title('Sinc signal');

% Signal xd - (Exponential signal)
xd = exp(-0.3*na);
subplot(224);stem(na,xd);
xlabel('n');ylabel('Amplitude');
title('Exponential signal');
```

(4) Sinusoidal Signal Generation (Sinu_Signal.m)

```
clear all;
close all;
clc;

La = 0.1;
f2a = 300;
f2b = 500;
fs2a = 5*f2a;
fs2b = 5*f2b;
t1 = [0:1/fs2a:0.1];
t2 = [0:1/fs2b:0.1];
x2a = sin(2*pi*f2a*t1); % Sinusoidal signal x1
x2b = sin(2*pi*f2b*t2); % Sinusoidal signal x2

subplot(121);stem(t1,x2a);
xlabel('Time (s)'); ylabel('Amplitude');
title('Sinusoidal Signal (300Hz) sampled at 5*300Hz');

subplot(122);stem(t2,x2b);
xlabel('Time (s)'); ylabel('Amplitude');
title('Sinusoidal Signal (500Hz) sampled at 5*500Hz');
```

(5) Audio Signal Generation (Signal_wavwrite.m)

```
clear all;
close all;
clc;

La = 0.1;
f2a = 300;
f2b = 500;
fs2a = 5*f2a;
fs2b = 5*f2b;
t1 = [0:1/fs2a:5];
t2 = [0:1/fs2b:5];
x2a = sin(2*pi*f2a*t1); % Sinusoidal signal x1
x2b = sin(2*pi*f2b*t2); % Sinusoidal signal x2

%Writing the data(audio signal) in .wav file
fs = 8000;
audiowrite('sig1.wav',x2a,fs);
audiowrite('sig2.wav',x2b,fs);

%Reading the .wav file
a1 = audioread('sig1.wav');
a2 = audioread('sig2.wav');

%Playing the sound signals
sound(a1,fs);
pause(2);
sound(a2,fs);
```

```
pause(3)

%Mixing of audio signals
a_mix = [a1;zeros(5000,1)] + a2 ;
sound(a_mix,fs);
```

#Exercise:

To combine two audio signals, they should have same dimension and same sampling rate. If dimensions are not same then zero padding needs to be done.

(6) Convolution (Signal_Conv.m)

```
clear all;
close all;
clc;

% Signal xa - (Rectangular signal)
La = 20; %signal length
na = [0:La-1]; %signal sample index
xn = [zeros(1,La/4) ones(1,La/2) zeros(1,La/4)];  %Rectangular signal

% Signal xc - (Sinc signal)
nb = [-La/2:La/2-1];
hn = sinc(0.2*pi*nb);

% Output of convolution
yn1 = conv(xn,hn);
[p,q] = size(yn1);
r = [0:q-1];

% Different plots
subplot(221);stem(na,xn);
xlabel('n');ylabel('Amplitude');
title('Rectangular signal, xn');

subplot(222);stem(nb,hn);
xlabel('n');ylabel('Amplitude');
title('Sinc signal, hn');

subplot(223);stem(r,yn1);
xlabel('n');ylabel('Amplitude');
title('Convolution of above two signals, yn1');

% Unit Impulse signal
deln = [ones(1,1) zeros(1,La-1)];

%Convolution of sinc and unit impulse signals
yn2 = conv(deln,hn);

figure
subplot(221);stem(na,deln);
xlabel('n');ylabel('Amplitude');
```

```
title('Unit Impulse signal, deln');

subplot(222);stem(nb,hn);
xlabel('n');ylabel('Amplitude');
title('Sinc signal, hn');

subplot(223);stem(r,yn2);
xlabel('n');ylabel('Amplitude');
title('Convolution of above two signals, yn2');
```

#Exercise:

(a) Number of elements in output of convolution = n1 +n1 – 1

(b) Convolution of any signal with unit impulse signal will be that signal itself.