

# ICS 311 Digital Signal Processing

## Lab8\_Audio signal

Name: Abhishek Harsh

2021BCS0036

1. Find 3 inbuilt audio signals in matlab repository and write a Matlab program to read an audio signal. Perform frequency sampling with 3 conditions ( $f_s > 2f_m$ ,  $f_s = 2f_m$ ,  $f_s < 2f_m$ ). Plot the graphs of the input and all the output signals.
2. Perform the sampling with above conditions on any song (song should not be more than 10sec length). Plot the input and the output graphs.

### Code:

```
% Abhishek Harsh
% 2021BCS0036
[x, fm] = audioread('example.mp3');
ly = length(x);
lspan = 1:ly;
t = lspan / fm;
figure;
% First subplot: Plot the waveform using the original time
subplot(2, 2, 1);
plot(t, x);
title('Original Waveform');
xlabel('Time (s)');
ylabel('Amplitude');

% Second subplot
fs1 = 1.5 * fm;
t1 = lspan / fs1;
plot(t, x, ':');
title('fs < 2fm');
xlabel('Time (s)');
ylabel('Amplitude');

% Define a new sample rate
F = 2 * fm;

% Third subplot
ts = lspan / F;
subplot(2, 2, 3);
plot(ts, x);
title('fs = 2fm');
xlabel('Time (s)');
ylabel('Amplitude');
% Define a new sample rate
```

```
F1 = 2.4 * fm;
```

```
% Third subplot: Plot the waveform using the new time vector
```

```
ts = lspan / F;
```

```
subplot(2, 2, 4);
```

```
plot(ts, x);
```

```
title('fs>2fm');
```

```
xlabel('Time (s)');
```

```
ylabel('Amplitude');
```

Output:

