Digital Signal Processing

Lab 4

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Assignment 3 :

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%%Assignment 3
%%Question 3:
seg =14;
sec = 42;
sampleRate = 44100;
t = linspace(0,sec,sec*sampleRate);
f =
[50,100,200,400,1000,2000,4000,6000,8000,10000,1200
0,14000,16000,18000];
for i=0:seg-1
    begin =(i*3*sampleRate);
    finish=(3*sampleRate)+begin;
    w =2*pi*f(i+1);
    signal(begin+1:finish) =
0.5*sin(w*t(begin+1:finish));
end
%length(t)
%length(signal)
%stem(t,signal);
%sound(signal,sampleRate)
audiowrite('file.wav',signal,sampleRate);
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%%Question 4:
cyc = 12;
f1 = 600;
f2 = 1400;
amp = 0.25;
sampleRate = 22050;
t= linspace( 0,cyc,cyc*sampleRate+1);
for i = 0:cyc-1
    begin = i*sampleRate;
    finish = (sampleRate/2)+begin;
    signal2(begin+1:finish) =0;
    w1 =2*pi*f1;
    w2 =2*pi*f2;
    signal2(finish : begin+sampleRate+1) =
amp*sin(w1*t(finish : begin+sampleRate+1))+
amp*sin(w2*t(finish : begin+sampleRate+1));
end
% length(t)
% length(signal2)
% stem(t,signal2);
%sound(signal2,sampleRate)
audiowrite('siren.wav',signal2,sampleRate);
```

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%%Question 5:
s = input('Please enter your telephone number: ',
's');
n = length(s);
sample_rate = 20000;
t = 1/sample_rate;
time_tone = [0:t:0.2];
time_silence= [0:t:0.05];
silence = zeros(1,numel(time_silence));
signal3 = [];
tx =[];

for i = 1:n
    switch s(i)
        case '0'
            f_low = 941;
            f_high = 1336;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 = [signal3 y_high+ y_low
silence];
        case '1'
            f_low = 697;
            f_high = 1209;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 = [signal3 y_high+ y_low
silence];
            tx = [tx time_tone time_silence];
        case '2'
            f_low = 697;
            f_high = 1336;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
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            signal3 =  [signal3 y_high+ y_low
silence];
            tx = [tx time_tone time_silence];
        case '3'
            f_low = 697;
            f_high = 1477;
            y_low= 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 = [signal3 y_high+ y_low
silence];
            tx = [tx time_tone time_silence];
        case '4'
            f_low = 770;
            f_high = 1209;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low
silence];
            tx = [tx time_tone time_silence];
        case '5'
            f_low = 770;
            f_high = 1336;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low
silence];
            tx = [tx time_tone time_silence];
        case '6'
            f_low = 770;
            f_high = 1477;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high =
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low
silence];
```

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            tx = [tx time_tone time_silence];
        case '7'
            f_low = 852;
            f_high = 1209;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high = 
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low 
silence];
            tx = [tx time_tone time_silence];
        case '8'
            f_low = 852;
            f_high = 1336;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high = 
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low 
silence];
            tx = [tx time_tone time_silence];
        case '9'
            f_low = 852;
            f_high = 1477;
            y_low = 0.15*sin(2*pi*f_low*time_tone);
            y_high = 
0.15*sin(2*pi*f_high*time_tone);
            signal3 =  [signal3 y_high+ y_low 
silence];
            tx = [tx time_tone time_silence];
    end
end
sound(signal3,sample_rate);
```

Assignment 4 :

```
%%Assignment 4:
comp_AXF("file.wav", "compressed.axf");

function comp_AXF(givenSound, filename)
    [givenSound , fs] = audioread(givenSound);

    samplingRate = 20000;
    givenSound = resample(givenSound,
samplingRate,fs);
    frame = 20000;
    minF = 400;
    maxF = 3400;

    no_frames = ceil(length(givenSound) / frame);
    pad_L = no_frames * frame;
    givenSound = [givenSound; zeros(pad_L -
length(givenSound), 1)];
    frames = reshape(givenSound, frame, no_frames);

    %BPF
    [b, a] = butter(4, [minF, maxF] * 2 /
samplingRate, 'bandpass');
    filter_frames = zeros(size(frames));
    for i = 1:no_frames
        filter_frames(:, i) = filter(b, a,
frames(:, i));
    end

    fft_frames = fft(filter_frames);

    time = length(givenSound) / samplingRate;
    save(filename, 'fft_frames', 'time',
'no_frames', 'pad_L');
end
```