C0E 472: DIGITAL SIGNAL PROCESSING

Lab 09: Encoding and Decoding Touch-Tone Signals

INDEX NUMBER: 3050420

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INTRODUCTION

This lab teaches how to design and implement bandpass FIR filters in MATLAB to decode Touch-Tone (DTMF) signals automatically. The main lab file is dtmfrun.m

The functions used: firfilt(), and freqz().

In the subsequent sections of the lab, we made four functions in .m files

Dtmfdesign.m, dtmfscore.m and dtmfcut.m, dtfmdial.m (found in appendix)

LAB EXERCISE - DTMF Decoding

4.1 Simple Bandpass Filter Design: dtmfdesign.m

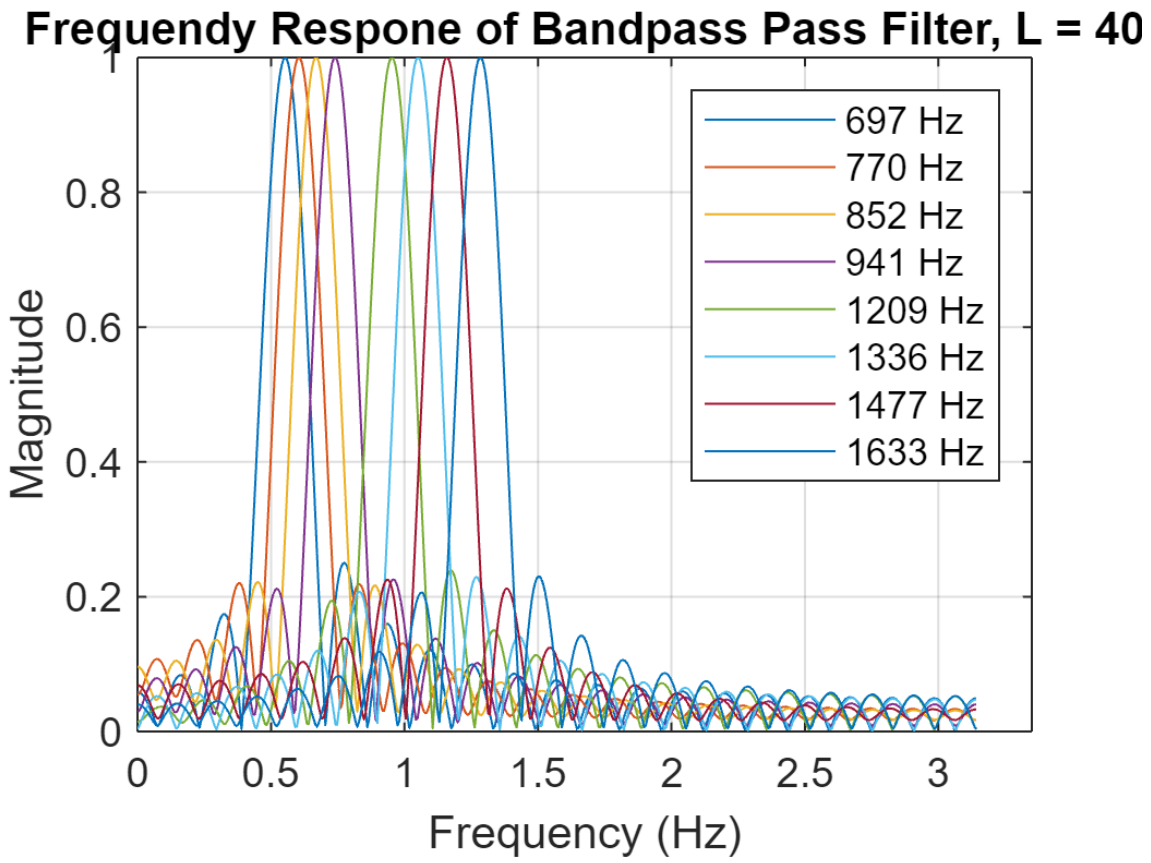
dtmfdesign function was made.

```
function hh = dtmfdesign(fb, L, fs)
    % dtmfdesign: Design DTMF bandpass filters
    % hh = dtmfdesign(fb, L, fs) returns a matrix (L by length(fb)) where each column
contains
    % the impulse response of a BPF, one for each frequency in fb.
    % fb = vector of center frequencies
    % L = length of FIR bandpass filters
    % fs = sampling frequency
    % Each BPF is scaled so that its frequency response has a maximum magnitude equal
to one.

    hh = zeros(L, length(fb)); % Pre-allocate hh matrix
    for k = 1:length(fb)
        nn = 0:(L-1);
        hh(:, k) = cos(2 * pi * fb(k) * nn / fs);
        ww = 0:pi/500:pi;
        HH = freqz(hh(:, k), 1, ww);
        zz = max(abs(HH));
        HH1 = freqz((1 / zz) * hh(:, k), 1, ww); % Scale to max value equal to one

        plot(ww, abs(HH1));
        xlim([0 3.35]);
        ylim([0 1]);
        grid on;
        hold on;
    end
    hold off;
end
```

d) Use dtmfdesign when L = 40 and fs = 8000Hz, using the eight DTMF Frequencies

```
centre_freq= [697 770 852 941 1209 1336 1477 1633];
a = dtmfdesign(centre_freq, 40, 8000);
title('Frequendy Respone of Bandpass Pass Filter, L = 40');
xlabel('Frequency (Hz)');
ylabel('Magnitude');
legend(arrayfun(@(f) sprintf('%d Hz', f), centre_freq, 'UniformOutput', false));
```

OUTPUT

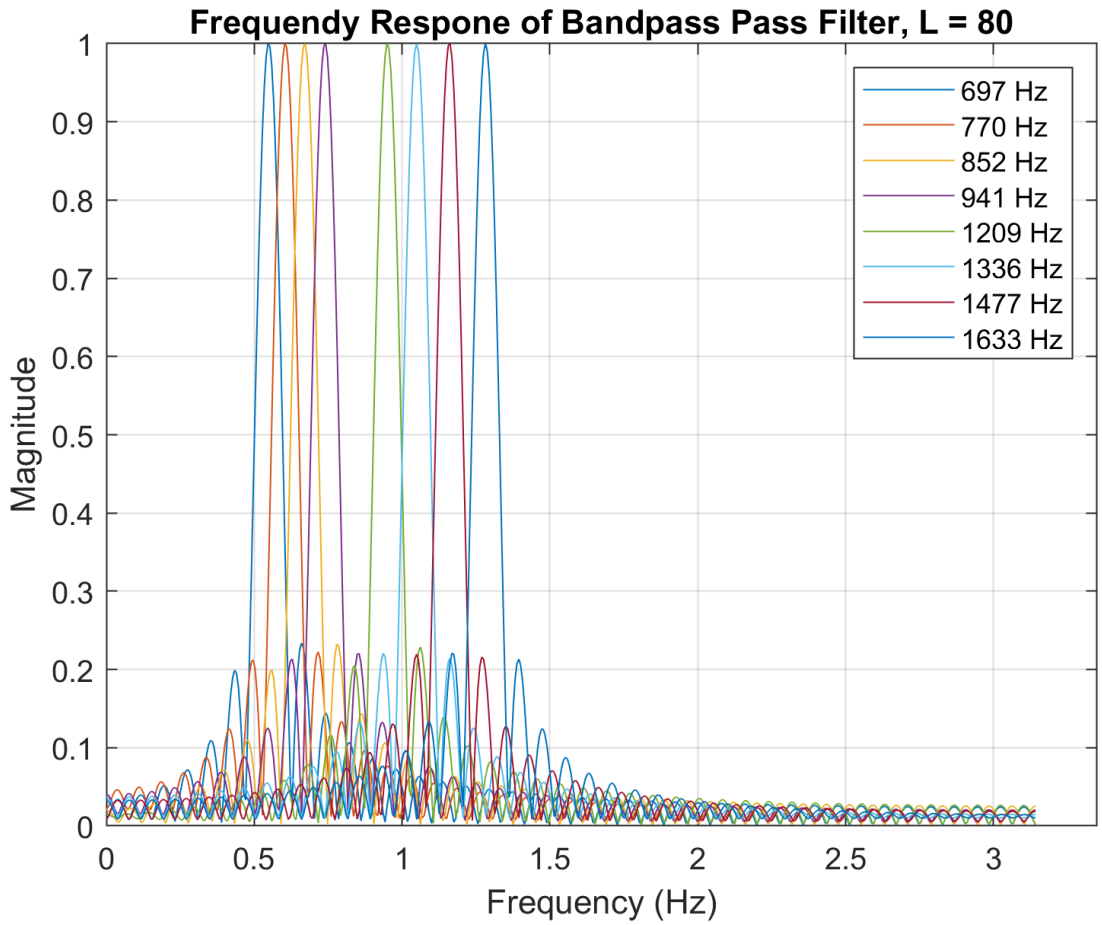


e) Use dtmfdesign when L = 80 and fs = 8000Hz, using the eight DTMF Frequencies

Code is the same as in d, except for the second line

```
a = dtmfdesign(centre_freq, 80, 8000);
```

OUTPUT



f)i. Find Minimum Length L

Section f has two parts

```
%(f)Finding minimum length L (part 1)
centre_freq = [697 770 852 941 1209 1336 1477 1633];
fs = 8000;
L = 2; % initial value for L
done = 0;


while (~done)
    k = 0;
    L = L + 1;
    hh = []; % initiate
    for i = 1:length(centre_freq)
```

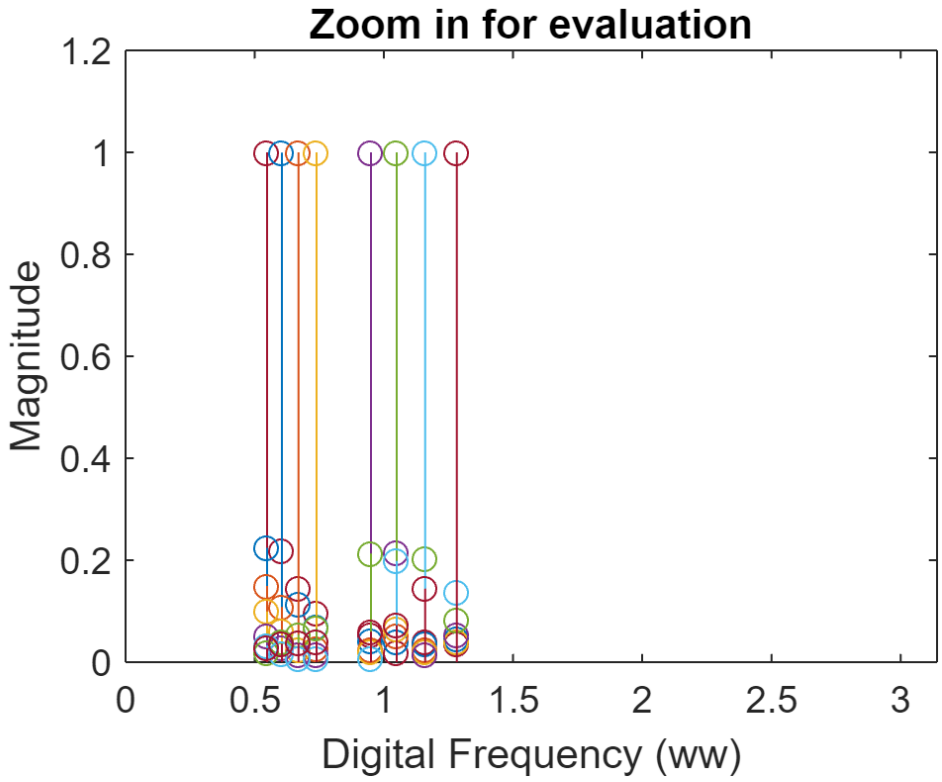
```
        nn = 0:(L-1);
        hh(:, i) = cos(2 * pi * centre_freq(i) * nn / fs); % filter impulse
response
        ww = 0:pi/1000:pi;
        HH = freqz(hh(:, i), 1, ww);
        HH1 = freqz((1 / max(abs(HH))) * hh(:, i), 1, ww);
        % frequency response of filter with maximum value equal to one
        for j = 1:length(centre_freq)
            z = round(2 * (centre_freq(j) / fs) * 1000);
            if abs(HH1(z)) < 0.25 % stop-band proportion 25%
                k = k + 1;
            end
            if k == 56
                done = 1;
            end
        end
    end
end


L = 88 % minimum L = 88
```

Thus L = 88

f)ii. Check if L satisfies the demand

```
f% (fii) Evaluation L whether satisfies the demand or not

L = 88;
centre_freq = [697 770 852 941 1209 1336 1477 1633];
w_loca = [1395 1541 1705 1883 2419 2673 2955 3267];
% the location for each eight frequencies' peak value
hh = zeros(L, length(centre_freq));
plot(0:0.001:pi, 0.25)
hold on
for i = 1:length(centre_freq)
    nn = 0:(L-1);
    hh(:, i) = cos(2 * pi * centre_freq(i) * nn / fs); % filter impulse response
    ww = 0:pi/8000:pi;
    HH = freqz(hh(:, i), 1, ww);
    HH1 = freqz((1 / max(abs(HH))) * hh(:, i), 1, ww);
    stem(ww(w_loca), abs(HH1(w_loca)));
    % stem the specific point for each eight filters
    hold on;
    title('Zoom in for evaluation')
    xlabel('Digital Frequency (ww)'); ylabel('Magnitude');
end
```

OUTPUT



L = 88

Comments on Selectivity:

- For each filter, I observed the passband at points where the magnitude is close to 1, and the stopband when the magnitude is less than 0.25. This selectivity ensures only one frequency component lies within the passband and the others lie in the stopband.

Analysis:

- With L=40, the bandwidth is wider, making it more challenging to isolate specific frequencies.
- Increasing L to 80 narrows the bandwidth, improving selectivity, but also increases the computational complexity.

Hardest Center Frequency to Meet Specifications:

- Filters at lower frequencies (e.g., 697 Hz, 770 Hz) face more challenges due to the closeness of frequencies, requiring a higher L for better isolation.

4.2 A Scoring Function

Creating the dtmfscore.m function.

```
%DTMFSCORE
% Usage: sc = dtmfscore (xx, hh)
% returns a score based on the max amplitude of the filtered output
% xx = input DTMF tone
% hh = impulse response of ONE bandpass filter
%
% The signal detection is done by filtering xx with a length-L
% BPF, hh, and then finding the maximum amplitude of the output.
% The score is either 1 or 0.
% sc = 1 if max(|y[n]|) is greater than, or equal to, 0.59
% sc = 0 if max(|y[n]|) is less than 0.59
%
function sc = dtmfscore (xx, hh)
xx = xx*(2/max(abs(xx))); %--Scale the input x[n] to the range [-2, +2]
yy=conv (hh, xx); % convolution of signal with BPF impulse response
if max(abs(yy))>=0.59 % binary output of signal presence in waveform
    sc=1;
else
    sc=0;
end
% figure; plot(abs(yy))
% Title ('Check for maximum amplitude function'), grid on
% xlabel('n'), ylabel('Magnitude')
end
```

Creating the dtmfcut.m function

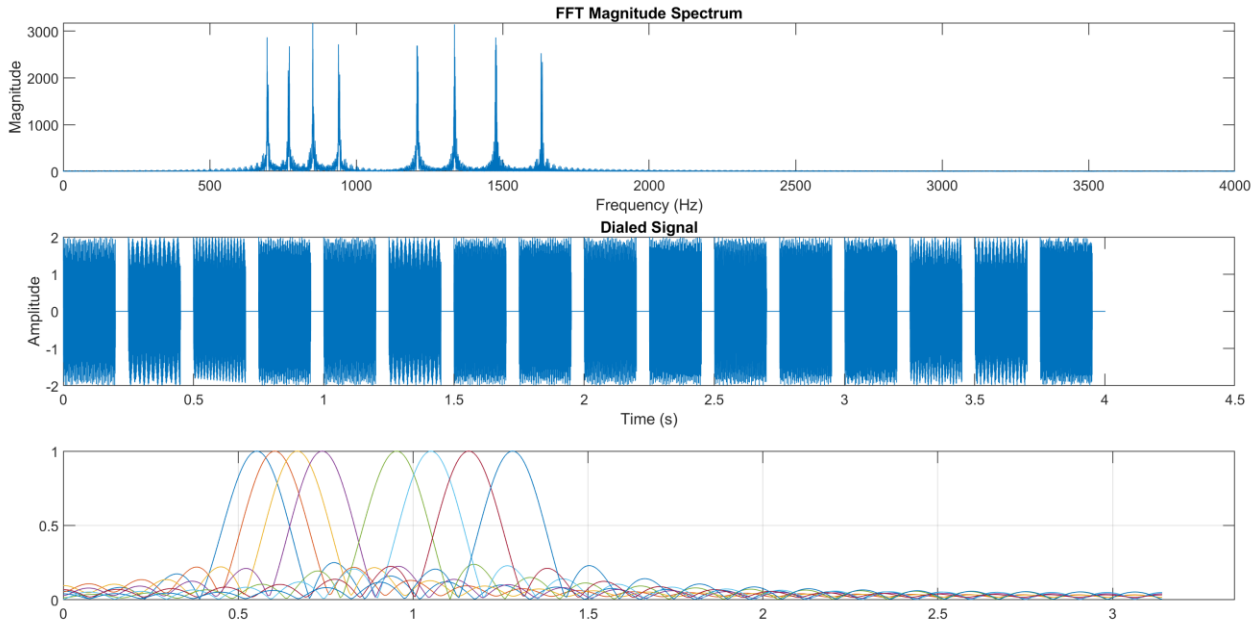
```
function [nstart,nstop] = dtmfcut(xx,fs)
%DTMFCUT   find the DTMF tones within x[n]
%  usage:
%       [nstart,nstop] = dtmfcut(xx,fs)
%
%   length of nstart = M = number of tones found
%      nstart is the set of STARTING indices
%      nstop is the set of ENDING indices
%    xx = input signal vector
%    fs = sampling frequency
%
%  Looks for silence regions which must at least 10 millisecs long.
```

```
%  Also the tones must be longer than 100 msec

xx = xx(:)'/max(abs(xx));   %-- normalize xx
Lx = length(xx);
Lz = round(0.01*fs);
setpoint = 0.02;      %-- make everything below 2% zero
xx = filter( ones(1,Lz)/Lz, 1, abs(xx) );
xx = diff(xx>setpoint);
jkl = find(xx~=0)';
%%%xx(jkl);
if xx(jkl(1))<0, jkl = [1;jkl];  end
if xx(jkl(end))>0, jkl = [jkl;Lx]; end
%%%jkl';
indx = [];
while length(jkl)>1
      if jkl(2)>(jkl(1)+10*Lz)
            indx = [indx, jkl(1:2)];
      end
      jkl(1:2) = [];
end
nstart = indx(1,:);
nstop = indx(2,:);
```

To check the code

```
xx=dtmfdial('1',8000); % for testing one filter at 697Hz
```

OUTPUT



d. Why the maximum value for $H(e^{j\omega})$ must be one for each bandpass filter
This guarantees predictable and consistent behavior, allowing accurate DTMF tone detection. This normalization ensures the filter neither amplifies nor attenuates the signal at the center frequency, facilitating reliable detection and consistent scoring.

4.3 DTMF Decode Function

Implement the dtmfrun.m function.

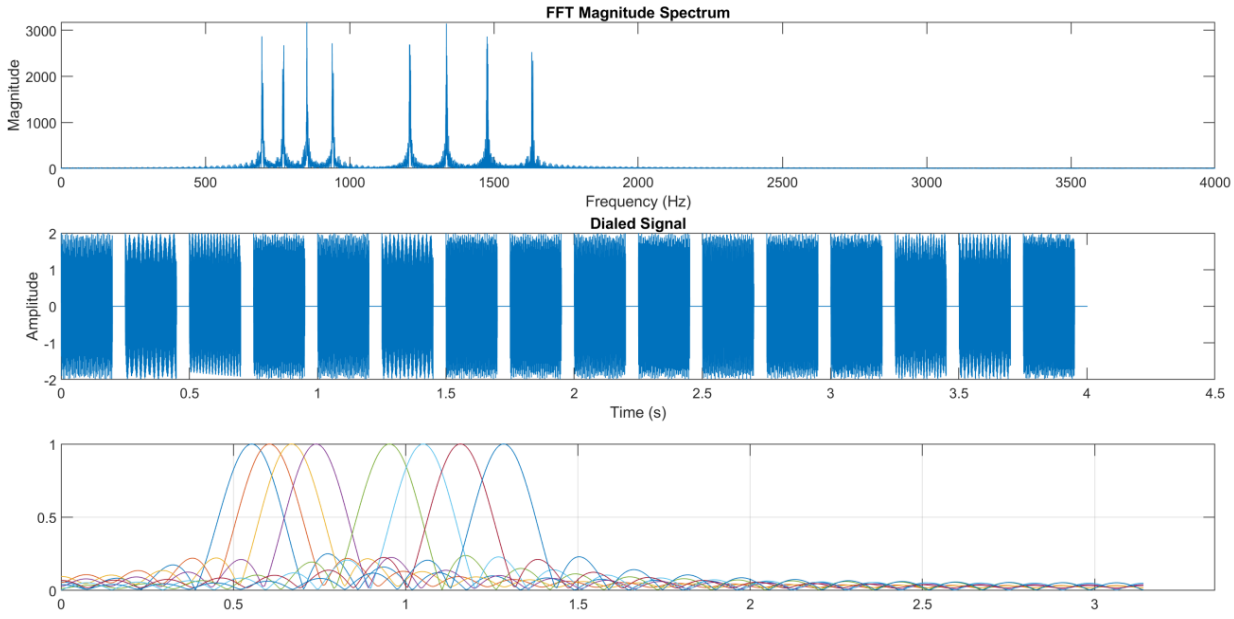
```
function keys = dtmfrun(xx, L, fs)
    center_freqs = [697 770 852 941 1209 1336 1477 1633];
    hh = dtmfdesign(center_freqs, L, fs);
    [nstart, nstop] = dtmfcut(xx, fs); % Find the beginning and end of tone
bursts
    xx = xx * (2 / max(abs(xx)));
    keys = [];
    dtmf.keys = ...
        ['1','2','3','A';
         '4','5','6','B';
         '7','8','9','C';
         '*','0','#','D'];


    % Uncomment and use this section if needed
    % for ii=1: length(nstart)
    %     x_seg=xx(nstart(ii): nstop(ii)); % Extract one DTMF tone
    %     score= [];
    %     for i=1:8
    %         score(i) = dtmfscore(x_seg, hh(:, i));
    %     end
    %     jkl=find(score==1);
    %     if length(jkl)~=2 % error indicator
    %         keys(ii)=-1;
    %     else
    %         yy=jkl(2)-4;
    %         keys(ii)=dtmf.keys(jkl(1), yy);
    %     end
    % end


    for kk = 1:length(nstart) % cycle through each tone
        n = [];
        x_1 = xx(nstart(kk):nstop(kk)); % Extract one DTMF tone
        for i = 1:length(center_freqs) % cycle through each filter
            zz = dtmfscore(x_1, hh(:, i));
```

```
            n = [n, zz]; % create a vector of ones and zeros representing where
the frequency components lie
        end
        aa = find(n == 1); % create a vector of indices where ones occur


        % Check for impossible scores and skip if they are found
        if length(aa) ~= 2 || aa(1) > 4 || aa(2) < 5
            keys = [keys, 'error'];
            continue
        end
        row = aa(1); % decode row position from aa
        col = aa(2) - 4; % decode column position from aa
        keys = [keys, dtmf.keys(row, col)]; % set keys equal to the current keys
and the key found in this iteration
    end
end
```

4.4 Telephone Numbers

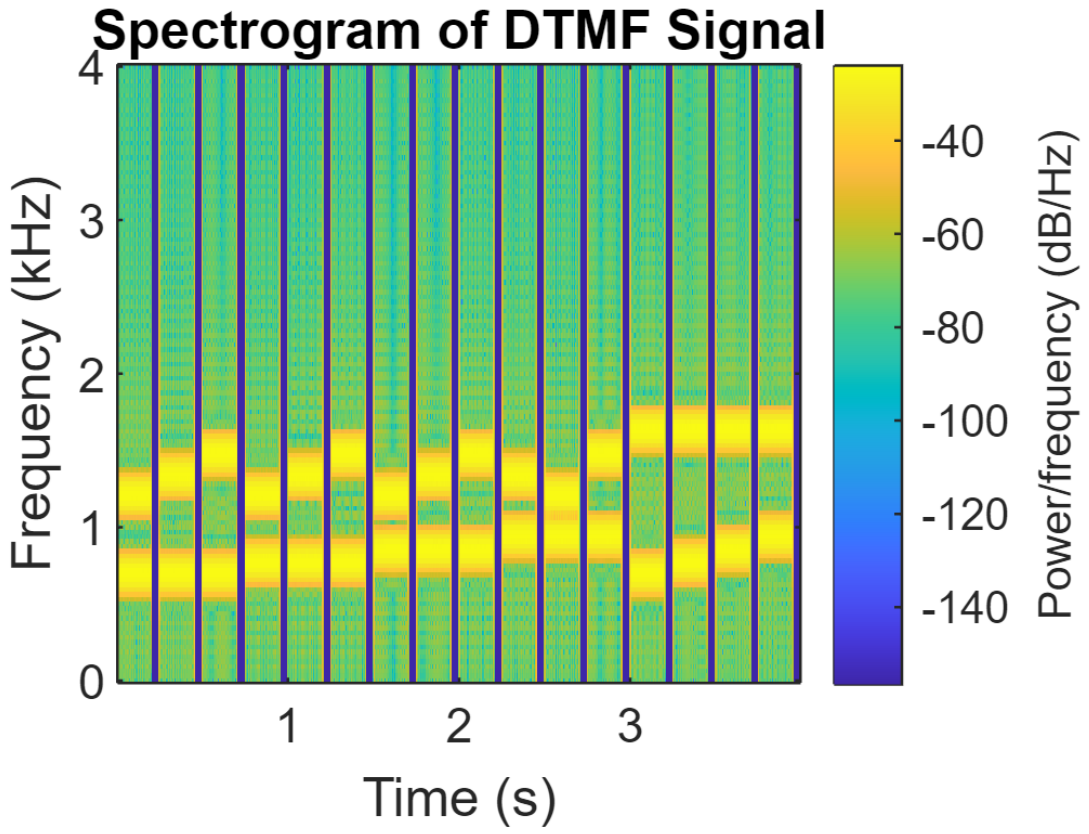
Test the complete system with a given phone number.

```
% Example usage:
fs = 8000;
L = 40;
tk = '1234567890*#ABCD';
xx = dtmfdial(tk, fs);
decoded_keys = dtmfrun(xx, L, fs);
disp(decoded_keys);
```



```
% Generate spectrogram for visualization
figure;
spectrogram(xx, 88, [], [], fs, 'yaxis');
title('Spectrogram of DTMF Signal');
```

OUTPUT



APPENDIX I: DTFMDIAL.M

```
function xx = dtmfdial(keyNames, fs)
    %DTMFDIAL Create a signal vector of tones which will dial
    % a DTMF (Touch Tone) telephone system.
    %
    % usage: xx = dtmfdial(keyNames, fs)
    % keyNames = vector of characters containing valid key names
    % fs = sampling frequency
    % xx = signal vector that is the concatenation of DTMF tones.


    dtmf.keys = ...
    ['1', '2', '3', 'A';
     '4', '5', '6', 'B';
     '7', '8', '9', 'C';
     '*', '0', '#', 'D'];
    dtmf.colTones = ones(4, 1) * [1209, 1336, 1477, 1633];
    dtmf.rowTones = [697; 770; 852; 941] * ones(1, 4);


    % Initialize the signal vector
    xx = [];


    % Duration of each DTMF tone pair
    toneDuration = 0.2; % 0.20 seconds
    silenceDuration = 0.05; % 0.05 seconds of silence between tones


    % Create each DTMF tone
    for k = 1:length(keyNames)
        key = keyNames(k);

        % Find the row and column for the key
        [ii, jj] = find(key == dtmf.keys);

        % Get the row and column frequencies
        f1 = dtmf.rowTones(ii, jj);
        f2 = dtmf.colTones(ii, jj);

        % Generate the time vector for the tone duration
        tt = 0:(1/fs):toneDuration;

        % Create the DTMF tone by summing two sinusoids
        tone = cos(2 * pi * f1 * tt) + cos(2 * pi * f2 * tt);

        % Concatenate the tone to the signal vector
```

```
        xx = [xx tone];

        % Add silence after the tone
        silence = zeros(1, round(silenceDuration * fs));
        xx = [xx silence];
    end

    % Plot the FFT magnitude spectrum
    N = length(xx);
    X = fft(xx);
    f = (0:N-1) * (fs/N); % Frequency vector
    figure;
    subplot(3,1,1);
    plot(f(1:N/2), abs(X(1:N/2)));
    title('FFT Magnitude Spectrum');
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');

    % Plot the dialed signal
    t = (0:N-1) / fs;
    subplot(3,1,2);
    plot(t, xx);
    title('Dialed Signal');
    xlabel('Time (s)');
    ylabel('Amplitude');

    % Plot the normalized frequency
    normalized_frequency = (0:N-1) * (2*pi/N); % Normalized frequency vector
    subplot(3,1,3);
    plot(normalized_frequency(1:N/2)/pi, abs(X(1:N/2)));
    title('Normalized Frequency');
    xlabel('Frequency (\pi rad/sample)');
    ylabel('Magnitude');
end
```