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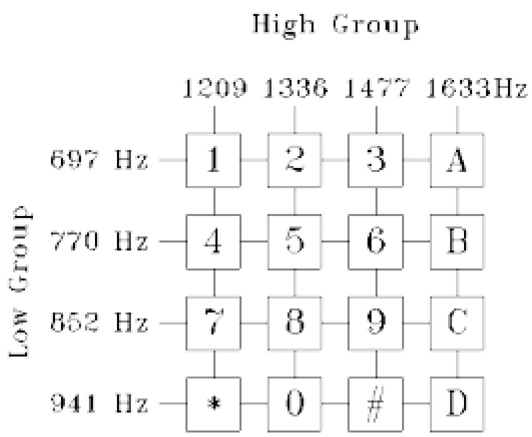
Experiment 3: Dual Tone Multi Frequency Encoder Decoder (DTMF)

AIM: To study and Analyse Coder/Decoder using FIR Filters in MATLAB

Dual Tone Multi Frequency is the basis for telephone system. Each key is mapped to a specified column and row frequency. As the key is pressed, two sinusoids (tones) are generated. These two tones are summed and this "Dual tone" signal now represents the pressed key. The two tones are carefully chosen so that no harmonics occur i.e No frequency is an integral multiple of the other and the sum/ difference of these two frequencies does not equal to any of the frequencies.

Eg: Pressing the digit 2 will generate the tones 1336 Hz and 697 Hz.



Decoding: The dual tone signal is now passed through a filter-bank containing 8 filters. Each filter is a band pass filter with the central frequency tuned to a tone and appropriately sharp cut-off to prevent overlapping with the adjacent tones. As the signal passes through the filter bank, the outputs of the filters tuned to the tones of the input signal are high in amplitude, the rest of the filter outputs are low amplitude outputs. With a simple thresholding scheme, the outputs are now 1s and 0s (digital output). Now this can be decoded using appropriate digital logic to get back the numbers.

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Code:

```
fr = [697 770 852  941];
fc = [1209 1336 1447 1633];
S='123A456B789C*0#D';
T=1;
t=0:T/10000:99999*T/10000;
y=zeros(1,8);
keys=zeros(1,10);
for i=1:10
    signal=sumsinusoid('4A*D8371*5',i,S,fr,fc);  %signal corresponding to
input key
    for j=1:4
        A=fir1(100,[fr(j)/5000-0.01 fr(j)/5000+0.01]);
        out=filter(A,1,signal);
        y(j)=any(out>1);
    end
    for j=1:4
        A=fir1(100,[fc(j)/5000-0.01 fc(j)/5000+0.01]);
        out=filter(A,1,signal);
        y(j+4)=any(out>1);
    end
    %decoder
    for j=1:4
        if(y(j))index=4*j-4;
        end
    end
    for j=1:4
        if(y(j+4))index=index+j;
        end
    end
    keys(i)=num2str(S(index));
end
keys=char(keys);


function signal=sumsinusoid(sequence,i,S,fr,fc)
T=1;
t=0:T/10000:9999*T/10000;

index=find(S==sequence(i));
temp=mod(index,4);
temp2=floor(index/4)+1;

if(~temp)
    temp=temp+4;
    temp2=temp2-1;
end

f1=fr(temp2);
f2=fc(temp);

signal=sin(2*pi*f1*t)+sin(2*pi*f2*t);
end
```

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Discussions:

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1. Due to inherent non-linearity of the channel, we have intermodulation distortion, which causes second and higher order terms in the spectrum of output. So, for successful decoding, we have to ensure that the two input tones are such that the output when band pass filtered does not give an erroneous decoded value. For this we ensure that the second and higher order terms do not coincide with any of the other frequencies. So, we choose f's such that no frequency is an integral multiple of the other and the difference or sum of any two frequencies should not equal any of the frequencies.

2. Here the sampling frequency is chosen as 10kHz, and as 10/2 > max(f)= 1633, it satisfies the Nyquist criterion and the band pass filtered output will be as desired. In practice, since the channel on which the dual tone is transmitted is the same as the audio channel (telephone line routed through operator), the sampling frequency will be the same as the standard for audio.

3. Depending upon the proximity of the frequencies, the required sharpness of bpf and threshold applied for decoding will change. If they are closer, we'll need higher order bpf and a higher thresholding for calculating binary y.

Saurabh Dash (14EC10050)

1. In this experiment while choosing the sampling frequency, it was ensured that the sampling frequency was greater than twice the maximum frequency, which in this case was 1633Hz.
2. We could have used on one tone per symbol but then that would lead to complication in the band pass filter design as the spectrum would now be more crowded and a sharper and greater order of filter would be required for decoding.
3. The frequencies for DTMF are chosen such that none of them have a harmonic relation with the others and that mixing the frequencies would not produce sum or product frequencies that could mimic another valid tone.