Experiment 4: DTMF

DSP Lab Assignment

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9th April, 2018

Dual Tone Multiple Frequency - Introduction

Dual-tone multi-frequency signaling (DTMF) is an in-band telecommunication signaling system using the voice-frequency band over telephone lines between telephone equipment and other communications devices and switching centers. DTMF was first developed in the Bell System in the United States, and became known under the trademark Touch-Tone for use in push-button telephones supplied to telephone customers, starting in 1963. DTMF is standardized as ITU-T Recommendation Q.23.It is also known in the UK as MF4.

The Touch-Tone system using a telephone keypad gradually replaced the use of rotary dial and has become the industry standard for landline and mobile service.

The engineers had envisioned telephones being used to access computers and automated response systems. They consulted with companies to determine the requirements. This led to the addition of the number sign (#, ''pound'' or "diamond" in this context, "hash", "square" or "gate" in the UK, and "octothorpe'' by the original engineers) and asterisk or "star" (*) keys as well as a group of keys for menu selection: A, B, C and D. In the end, the lettered keys were dropped from most phones, and it was many years before the two symbol keys became widely used for vertical service codes such as *67 in the United States of America and Canada to suppress caller ID.

Public payphones that accept credit cards use these additional codes to send the information from the magnetic strip.Present-day uses of the A, B, C and D signals on telephone networks are few, and are exclusive to network control. For example, the A key is used on some networks to cycle through different carriers at will. The A, B, C and D tones are used in radio phone patch and repeater operations to allow, among other uses, control of the repeater while connected to an active phone line.

The *, #, A, B, C and D keys are still widely used worldwide by amateur radio operators and commercial two-way radio systems for equipment control, repeater control, remote-base operations and some telephone communications systems.DTMF signaling tones can also be heard at the start or end of some VHS (Video Home System) cassette tapes. Information on the master version of the video tape is encoded in the DTMF tone. The encoded tone provides information to automatic duplication machines, such as format, duration and volume levels, in order to replicate the original video as closely as possible.

DTMF tones are used in some caller ID systems to transfer the caller ID information, but in the United States only Bell 202 modulated FSK signaling is used to transfer the data.

Keypad and Generation

The DTMF telephone keypad is laid out in a 4×4 matrix of push buttons in which each row represents the low frequency component and each column represents the high frequency component of the DTMF signal. Pressing a key sends a combination of the row and column frequencies. For example, the key 1 produces a superimposition of tones of 697 and 1209 hertz (Hz). Initial push button designs employed levers, so that each button activated two contacts. The tones are decoded by the switching center to determine the keys pressed by the user.

The DTMF keypad with the frequencies is shown below.

DTMF keypad frequencies (with sound clips)

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

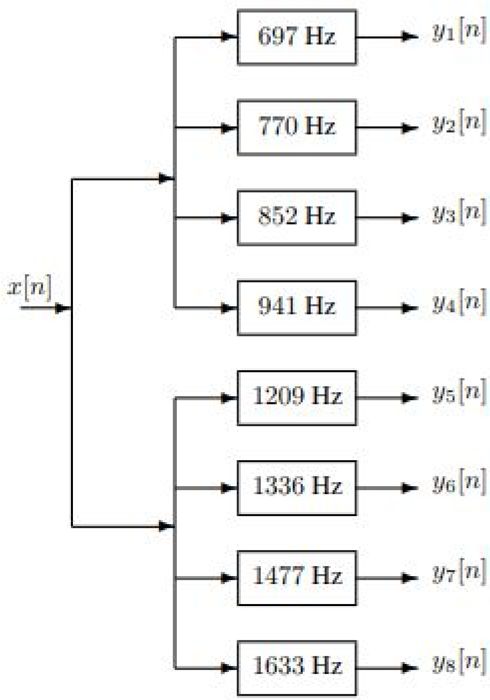
A DTMF tone representing a single key press on a telephone device consists of two summer frequencies that have been chosen such that no harmonics occur i.e., no frequency is a integral multiple of another and the sum or the difference of any two frequencies does not equal any of the frequencies.

Decoding

DTMF was originally decoded by tuned filter banks. By the end of the 20th century, digital signal processing became the predominant technology for decoding. DTMF decoding algorithms typically use the Goertzel algorithm. As DTMF signaling is often transmitted in-band with voice or other audio signals present simultaneously, the DTMF signal definition includes strict limits for timing (minimum duration and interdigit spacing), frequency deviations, harmonics, and amplitude relation of the two components with respect to each other.

Typically, a filter bank is used to detect the frequencies associated with a DTMF signal. The filter bank consists of 8 band pass filters, each of which passes only one of the 8 possible DTMF frequencies. The input signal for all filters is the same DTMF signal. When the input to the filter bank is a DTMF signal, the outputs from two if the band pass filters should be larger than the rest.

If we detect or measure which two outputs are the larger ones, then we know the corresponding frequencies, These frequencies are then used as row and column pointers to determine the key from the DTMF code. A good measure of the output levels is the peak value at the filter outputs. As when the band pass filter is working properly, it should pass only one sinusoidal signal and the peak value would be the amplitude of the sinusoidal passed by the filter.



Filter Bank Consisting of 8 Band Pass Filters.

Band Pass Filter Design

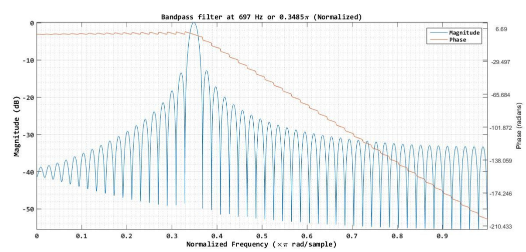
The L-Point filter design can be used to design a bandpass filter. The Lipoint average filter is a low pass filter that is primarily used to design low pass filters. Its pass bandwidth is inversely proportional to L. It is also possible to modify the design to obtain a band pass filter centered around some frequency other than zero. One simple way to do this is to define the impulse response of the L-point FIR Filter as

$$h[n] = \beta\cos(\hat{w}c*n) \text{ where } 0 \le n < L.$$

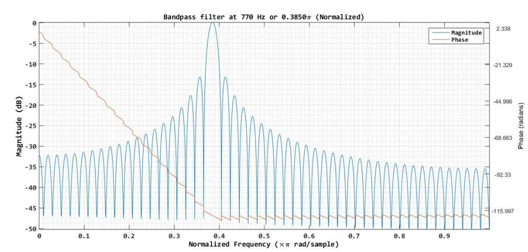
where L is the filter length of the FIR filter, $\hat{w}_c$ is the center frequency that defines the frequency location of the pass band and $\boldsymbol{\beta}$ is used to adjust the maximum value of the frequency response magnitude to be 1. The bandwidth of the bandpass filter is controlled by L, larger the value of L, narrower the bandwidth. In our experiment the L is taken to be 101 or the order is kept 100.

Frequency Response Of the 8 Band Pass Filters with Sampling frequency 4000 Hz

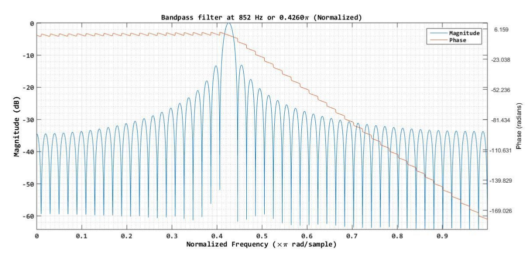
The Low Frequency Filters



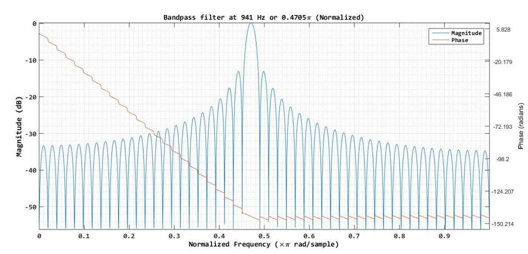
Band Pass Characteristics for 697 Hz



Band Pass Characteristics for 770 Hz

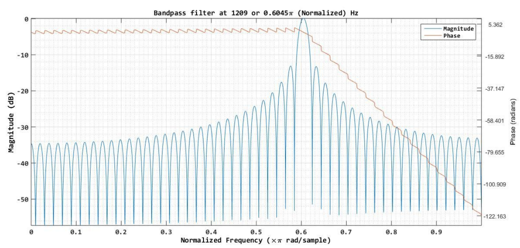


Band Pass Characteristics for 852 Hz

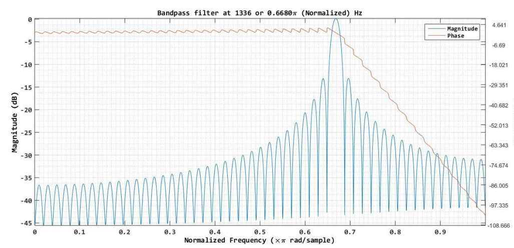


Band Pass Characteristics for 941 Hz

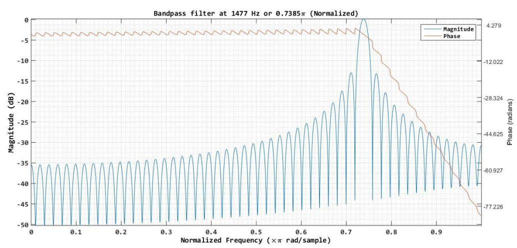
The High Frequency Filters



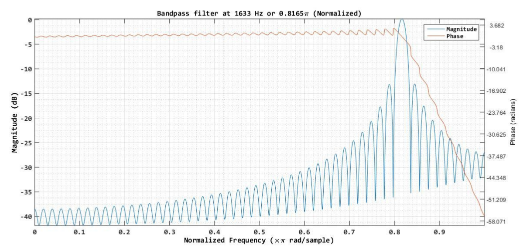
Band Pass Characteristics for 1209 Hz



Band Pass Characteristics for 1336 Hz



Band Pass Characteristics for 1477 Hz



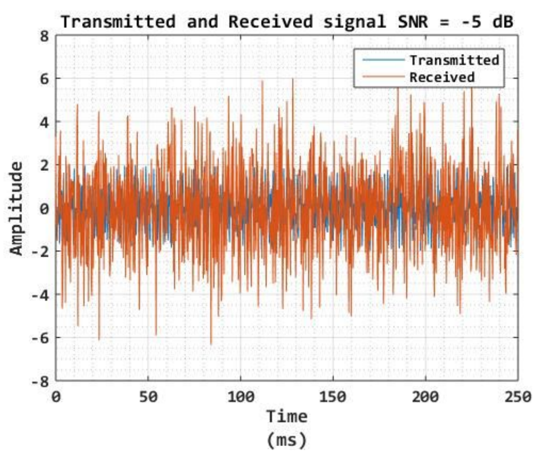
Band Pass Characteristics for 1633 Hz

The bandpass filters were designed to have the first side lobe attenuation greater than 10 dB. To achieve the maximum magnitude of the frequency response to be 1(or 0 dB), the value of β was varied. At β=0.2 we obtain the maximum magnitude of the frequency range along all frequency range to be 1 and the minimum attenuation of the side lobes to be 10 dB.

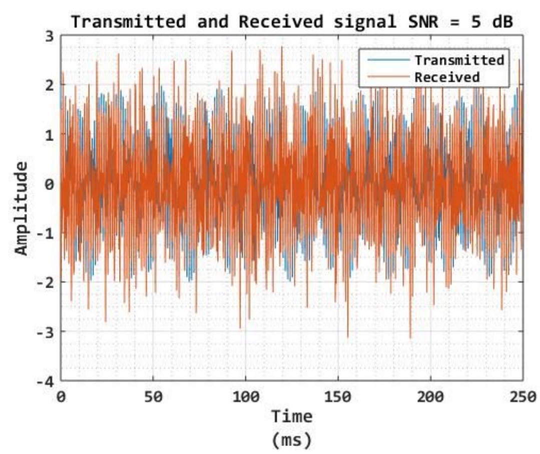
Objective

Generate 128 random symbols consisting only ['1', '2', '3', '4', '5', '6', '7', '8', '9', '*', '0', '#', 'a', 'b', 'c', 'd']. Depending on the symbol presses on the keypad, 2 tones are generated and combined to form $x(t)$ for each symbol for a fixed time duration of 250 ms.The 8 Bandpass filters of above cutoff frequencies were designed earlier were used. Then some AWGN noise was added to $x(t)$ and then retrieved the symbols by filtering the signal $y(t)=x(t)+n(t)$ where $n(t)$ is the noise component. This experiment was repeated 100 times for a particular SNR and the average number of error symbols for some SNR were observed.

Plot of Transmitted and Received Signals



Transmitted And Received Signal SNR -5 dB



Transmitted And Received Signal SNR +5 dB

Observation

The noise increases as we decrease SNR value and thus it will lead to increased misclassification of symbols.

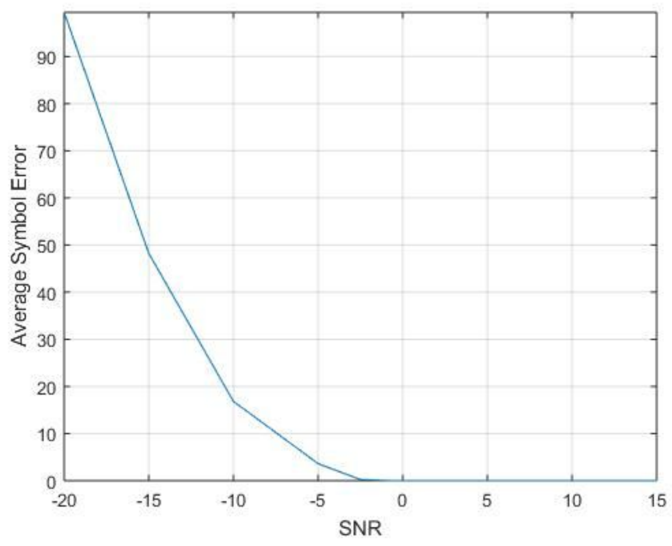
Results

After 100 iterations of each case,

128 symbols are generated in each iteration. The average symbol error is the number of mismatching symbols in 1 iteration.

SNR (dB)	Average Symbol Error
-20	99.37
-15	48.23
-10	16.82
-7.5	10.23
-5	3.61
-2.5	0.24
-1	0.03
-0.9	0.02
-0.8	0.01
-0.75	0.00
-0.5	0.00
0	0.00
1	0.00
5	0.00
15	0.00

Variation of Average Symbol Error with SNR



Average Symbol Error v/s SNR

Discussions

The dual tone multi frequency transmission, increases the robustness of the system.From the results we can see, the transmission is error free from SNR = −0.80 dB or more. So minimum SNR for this case is found −0.80 dB for error free transmission. The bandpass filter we used is simple FIR filter by using windowing, but we can use more complex higher order filters. The frequencies should be chosen very carefully, so that the signals can be filtered properly. The symbol time can be varied, but we need more sampling frequency for smaller duration, so that we have enough data.The error increases as SNR decreases.