SSY130, Applied Signal Processing

Project 1B Acoustic Communication System

Group 8 Secret Key: Electivire DOB: 19880803

Kashif Shabir Haitham Babbili Muhammad Haris Khan

Task 1:

The sampling frequency $f_s = 16$ Khz after interpolation of factor 8 means that the f_s before interpolation would be 16/8=2KHz, which means our signal starts from -1KHz to 1KHz. After modulation at frequency=4Khz the range of frequencies will be from 3KHz to 5KHz.

Task2:

EVM is a measure of how far the received points from the ideal constellation points. If SNR is a finite number instead of infinity the received points will always show some deviation from the ideal constellation points due to the added random noise since we have limited power availability, which means EVM will always show some value even for ideal channels. The reason why EVM always is nonzero is caused by the low-pass filter not being ideal which in turn means that the reconstruction of the signal is not perfect

Task 3:

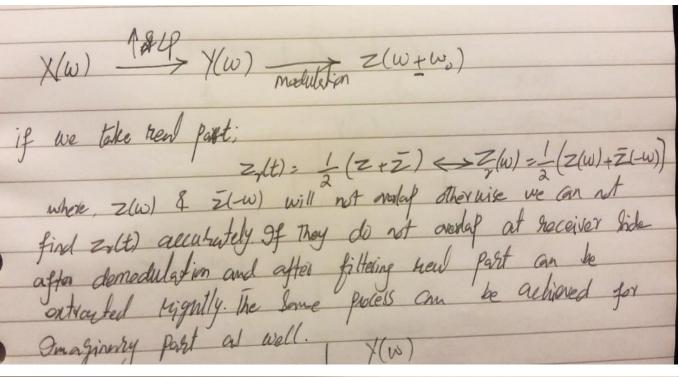
The channel H is estimated using the pilot OFDM blocks. H is effected by propagation over the physical channel and real part of the signal. The propagation over physical channel affects the signal in magnitude and phase (ideally) but affects in lot of other ways in addition to magnitude and phase if channel is non ideal.

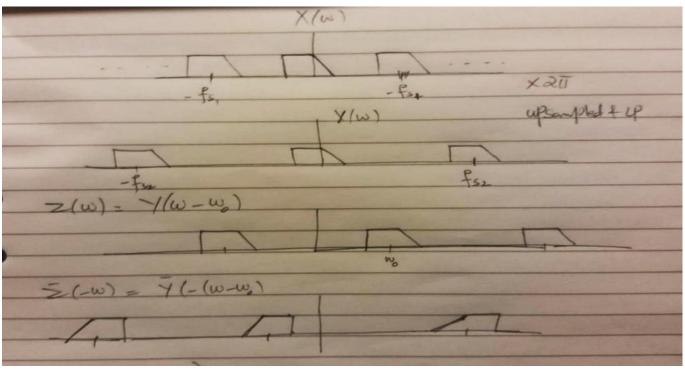
Similarly, real part is affected same as imaginary part under the assumption that our channel will affect both similarly since our supposed channel only accepts real signal we will append real part into the imaginary part to achieve a complex signal. Another way is to look in frequency domain

$$Z_r(w) = \frac{1}{2}(Z(w) + \bar{Z}(w))$$

Furthermore, H will not be affected by interpolation and decimation if Nyquist criteria fulfils. And modulation and demodulation will also not affect H because their functionality is reversible.

Task 4:





Task 5:

We found that Low pass filter characteristics act equal importance for interpolation and decimation stages to remove aliasing affects in the receiving side [1]. Therefore, all the four factors are important in the same way for both stages.

- 1. The phase linearity is at top for interpolation and decimation, since having non linearity we cannot estimate o/p accurately because the signal's shape changes rapidly, we can't preserve the signal's components and if we can't perverse signal then passband ripple, stopband attenuation and transition band width became of no use.
- 2. The transition bandwidth takes the 2nd position for both stages, because this attenuation can cause ISI because it widened the received signal and its less important than 1 because ISI can be eliminated by using other techniques but signal's modified shape cannot be recovered easily.
- 3. Both passband ripples and stopband attenuation are equally important for both stages, because they are linked with an inverse relation. Decreasing one will increase other. However, there importance can be defined having the prior specified filter characteristics, then we can design the filter using whose requirements.

Part #7
In order to prove that $H = \overline{T}R$ & $\widehat{H} = \overline{T}R$ equal we are going to prove in a way that both equations satisfy $L \cdot H \cdot S = R \cdot H \cdot S$
$H = \overline{T}R$ $H = \overline{T}'R$
$R = \frac{H}{T} = \hat{H}(\bar{T})^{T}$ $X = TH$ $X = TT R$ $X = TT R$
$(\overline{T})H = \overline{T}.(\overline{T})R \qquad TH = IR : T\overline{T}'_{-I}I$
$R = IR$ $R = IR \Rightarrow R = R$
R = R
Hence both equations satisfy that L.M.s = R.M.s
which means both are finisher
Now for magnitude & Phase,-
Magnitude: $ H = T /R!$ $ H = T' /R!$
$\frac{1H}{1R1} = 1\overline{7}1$ $\frac{1H}{1R1} = \overline{7} $
Similary *H-*R= *7 1 *H-*R= *7'
Hence, mignitude & phase of both cases are equal if $1.511 - 1.711$ $8 \le H = \le \hat{H}$.
311 - 411.

Task 8:

Par	18
Reg = R. A	Per = R/H
* Reg = * R + * A	= R. H ⁻¹
: \$ A = \$ T + \$ R	FRES = FR-FH
3 A = - 4T - 4R	: 3H = 3T+ 3R
* Rey = \$R \$ - \$R - \$T	* Reg - * R - * T' - * K
* Rg = - * 7	4 Reg = - ≠ T'
Since we already implies that Phase of	froved that $T = (T)$, which equalization is some for A & M
J	

Task 9:

With higher volume the SNR also increases which improves the results. When we started changing the signal amplitude from low to high the constellation points around transmitted point starting shrinking and making a more compact cloud and EVM value also stated decreasing since EVM is a measure of how far the points from the reference point. It is because that when we increase the signal amplitude we actually increase power which means we are increasing SNR. Higher the SNR greater will be the reliability of detection of the symbols. Same is the case with EVM because it also decreased means shrinking towards reference point because of added more power.

Task 10:

When we started moving our transmitter towards and away from the kit, we found that moving the speaker changes the transmission time of the pilot and data shortening the channel length and EVM generally makes a parabola. Since more power means lower EVM and vice versa. And this parabolic behavior can be explained with the help of OFDM. Since OFDM carries more carriers which means multipath effect will decide the power of the signal. These multipath effect defines constructive and destructive behavior. Now, at higher *d* between transmitter and kit we got higher EVM because of the destructive behavior and far away from the kit, but at some point when we move it will give us lowest EVM because at that point constructive behavior enhances the power significantly that's why EVM is minimum but when we move more towards the kit it again starting behaving like destructive.

Speed of the movement can be calculated using the constellation diagram by extracting the time delay t_d using cross correlation between transmitted and received symbols and then and distance can be calculated using

$$d = \frac{t_d}{f_s} C$$

Where c is speed of sound and f_s is the sampling frequency. Using distance one can find the speed of the movement by d/t. Sound-waves travel at the speed of sound, vs = 343 m/s. From basic physics, the distance traveled during the time delay should be $\Delta d = t0$ · vs. The pilot and message is sent in $(16+8)\cdot 4\cdot 8\cdot 2 = 1536$ symbols which in turn gives us the time it takes for the whole message to be transmitted equal to $\Delta t = 1536$ fs = 0.096s The velocity of the speakers will then be $V = \Delta d$ $\Delta t \approx 5.6$ cm/s. If the phase shift extends π 4 rad then for a QPSK constellation the symbols will be mapped incorrectly. Thus leading to a maximum velocity, when using 4kHz as the frequency, vmax \approx 8.9cm/s.

[1] Page 18, "project detail".