



CHALMERS

CHALMERS UNIVERSITY OF TECHNOLOGY

APPLIED SIGNAL PROCESSING

Project-1 Part-B Report

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1 Project description

The project is to design a data transmission system over an acoustic channel transferring text messages between two units via sound in the air. Part B of the project addresses sections 4 and 5 in the following:

1. Signal encoding/decoding from the bit stream to symbols
2. Conversion to an analogue coding/decoding technique; orthogonal frequency division multiplexing (OFDM).
3. Transmission channel equalization.
4. modulation/demodulation of the signal to/from a frequency band suited for acoustic transmission.
5. interpolation/decimation to convert the signal between different sampling rates in the system.

2 Findings and observations for Matlab implementation

2.1 Question-1

Assume the baseband signal is interpolated by a factor of 8, a modulation frequency of 4 kHz is used, and end up at a final sampling frequency of 16 kHz. What is the bandwidth of the transmitted audio signal and between which frequencies does the transmission band lie?

Answer: In this case, it is given that: Sampling frequency, $f_s = 16\text{kHz}$, modulation frequency = 4kHz and interpolation, $R = 8$. Therefore, after upsampling, we get $f_s/R = 16\text{kHz}/8 = 2\text{kHz}$. which denotes that the signal lies between -1kHz and 1kHz across the origin. Once we modulate this signal at 4kHz the signal gets shifted to 4kHz such that it lies between 3kHz i.e $(4+(-1\text{kHz}))$ and 5kHz i.e $(4+1\text{kHz})$ such that the bandwidth will be 2kHz i.e $(3\text{kHz}-5\text{kHz})$.

2.2 Question-2

In part A we could achieve an EVM which was nearly zero if we set $\text{snr}=\text{inf}$. However, if we try to do the same thing with our new setup, we always get a nonzero EVM. Why is this the case?

Answer: During transmission, we are only transmitting the real part of the signal and we are regenerating the complex at the receiver end such that the exponential part can

vary at the receiver end. Hence this can result in non-zero EVM.

Also at the interpolation and decimation stages, the filter is not ideal so the truncation is not perfect and this can also result in non-zero EVM.

2.3 Question-3

In the receiver, H is estimated from the pilot OFDM block. What parts of the system (i.e. interpolation, modulation, taking the real part of the signal, propagation over the physical channel, demodulation, decimation) contribute to channel H ? How do they contribute?

Answer: Interpolation process include up-sampling $z(n)$ by R followed by an ideal low-pass filter with a normalized cutoff frequency of $1/R$. Thus it is the process of increasing the sampling frequency of a signal to a higher sampling frequency. A low pass filter is used to eliminate the copies. If interpolation is not present there will be a chance of having combined positions of the plotted signal, which results in getting the wrong effective channel impulse response since it is calculated using the received pilot data.

During modulation, we multiply each sample with a complex exponential with shifts the absolute position of a signal in the frequency domain. This is done in order to convert the signal from baseband frequency to passband frequency so that the signal can be transmitted through the channel. Since this modulation is done to the signal which contains both pilot and data, the estimation of the channel impulse response is highly affected.

Signals send over a medium is always real. Also, neither the magnitude of the signal nor the phase contains any imaginary parts. At the receiver end, we will be able to regenerate the complex part once we retrieve the real part of the signal. Thus We transmit only the real part of the signal.

Propagation of the signal over the physical channel will always result in distortion of the signal in the non-ideal cases due to the AWGN noise which results in a variation of amplitude, and phase and this may result in bit error, synchronization error, delay or loss of signal. Thus propagation can very badly affect channel estimation due to the above reasons.

During demodulation, we nullify the effect of modulation by multiplying with the same complex exponential with sign-reversed and thus convert the signal from the passband frequency to the baseband frequency, which is the inverse of modulation.

Decimation is the inverse process of interpolation where we first apply an anti-aliasing low pass filter to the demodulated signal. Then we reduce the sampling rate by only

keeping the samples of the original signal and discarding the rest.

2.4 Question-4

Before transmission over the physical channel we have chosen to simply discard the imaginary part of the complex-valued signal. Explain why we can safely discard either the imaginary part of the signal.

Observation and inference: During transmission we are transmitting only the real part of the signal and the complex part is discarded. if we consider the complex signal for example as $a+ib$, when we say we are transmitting only the real part, it means that we are transmitting the signal with a and b . Once we do the Fourier transform, the magnitude and phase do not contain any imaginary part. since we have both a and b received at the receiver, we will be able to regenerate both the real and complex parts since one is the conjugate of the other.

ex: Let's take $Z(\omega) = a + jb = f_1(a, b)$.

$$Z_r(\omega) = \frac{1}{2}[Z(\omega) + \bar{Z}(-\omega)] = f_2(a, b)$$

$$\text{Re}(Z) = Z_r = |Z_r| \exp j\phi_{Z_r} \text{ where } |Z_r| = f_3(a, b) \text{ and } \phi_{Z_r} = f_4(a, b)$$

here we transmit the amplitude and phase. Therefore the real (a) and imaginary (b) part of the Z is preserved while transmitting.

2.5 Question-5

In the interpolation and decimation stages, which of the following lowpass filter properties are more important, and which are less important? Why? Specify your answers separately for the interpolation and decimation stages. Passband ripple, Stopband attenuation, Transition bandwidth and Phase linearity

Observation and inference:

-Passband ripples are observed in the high gain region of the higher order filter. Less passband ripples result in higher stopband attenuation.

-Stopband attenuation is the difference (in dB) between the lowest gain in the passband and the highest gain in the stopband. Higher stopband attenuation is better.

-Transition bandwidth is the range of frequencies which allows a transition between the passband and the stopband of a filter. The higher the stopband attenuation, the smaller the transition bandwidth and this reduces the aliasing of the signals.

-Phase linearity is a property of the filter where the phase response of the filter is a linear function of frequency. If the phase change is within the decision region, we will be able to recover the signal. Hence, it is comparatively less important compared to the rest of the properties.

At the interpolation stage, the transition bandwidth has the most importance since the higher the stopband attenuation, the lesser the ripples in the stop band and this in turn reduces the transition bandwidth. If the transition is gradual and its bandwidth is lower, the interpolation will be almost ideal.

At the decimation stage, the transition bandwidth and stopband attenuation has the most importance since distortion caused due to aliasing is very less if the stopband attenuation is higher as it reduces the transition bandwidth to a lower value.

3 Findings and observations for the DSP-kit tests

3.1 Question-6

The channel estimate is generated as $\hat{H} = \bar{T}R$

We avoid using the division operation because it is computationally much more demanding than performing a complex conjugation and multiplication. Assuming an ideal system, show that \hat{H} and H have the same phase and magnitude.

Observation and inference: The estimated channel response is given by the equation,

$$\hat{H} = \bar{T}R$$

In the above channel estimation formula we are multiplying R by complex conjugate of T instead of dividing R by T in order to reduce the complexity of computation.

Here we assume system is ideal, then $phase(\bar{T})$ will be equal to $-phase(T)$ thus the phase difference between pairs (R, \bar{T}) and (R, T) will be the same. Thus, $phase(\hat{H}) = phase(H)$. Even

though, the magnitude of the \hat{H} will be scaled but later it will be normalized to unit magnitude. Thus $\text{magnitude}(\hat{H}) = \text{magnitude}(H)$.

Hence, we can conclude that the phase and magnitude of channel response and estimated channel response in an ideal system will be the same.

3.2 Question-7

The received symbols are equalized as $\text{Req} = \mathbf{R} \cdot \overline{\hat{H}}$. Show that this correctly equalizes the phase of Req.

Observation and inference: $\text{Req} = \mathbf{R} \cdot \overline{\hat{H}}$

Thus the $\text{phase}(\text{Req})$ will be the sum of the $\text{phase}(\mathbf{R})$ and $\text{phase}(\overline{\hat{H}})$.

We know that $\text{phase}(\overline{\hat{H}})$ will be $-\text{phase}(\hat{H})$. Such that the $\text{phase}(\text{Req})$ will become equal to $\text{phase}(\mathbf{R})$. Thus $\text{phase}(\text{Req})$ is correctly equalized.

3.3 Question-8

With a constant background noise level and physical setup, test sending messages with high and low signal amplitudes. (Use the serial monitor to change the amplitude in real time). How do the EVM and constellation diagram change? Why?

Observation and inference: We can observe that the received constellations are tending outwards from the middle (0,0) when increasing the amplitude. EVM is reduced until the amplitude is increasing up to the amplitude of transmitting. While keep increasing, again EVM is increased.

Generally, we can see that the receiving constellations are spreading outward of the root. While performing the test for several times in each low, mid, and high amplitude level, we can observe more incorrect messages at low amplitudes than at high amplitudes. [Figure 1 and 2 and 3]



Figure 2: Constellation diagram for Mid amplitude

