SSY130 Applied Signal Processing

Project 1B: Acoustic Communication System

Group 9

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I. Introduction

In part B, the channel in FIG. 1 consists of the loudspeaker/microphone, and the interpolation, modulation, demodulation, and decimation steps. We simulate the effect of performing interpolation, modulation, demodulation, and decimation in *Matlab*. Then we will test OFDM on the DSP-board and try to send short text messages.

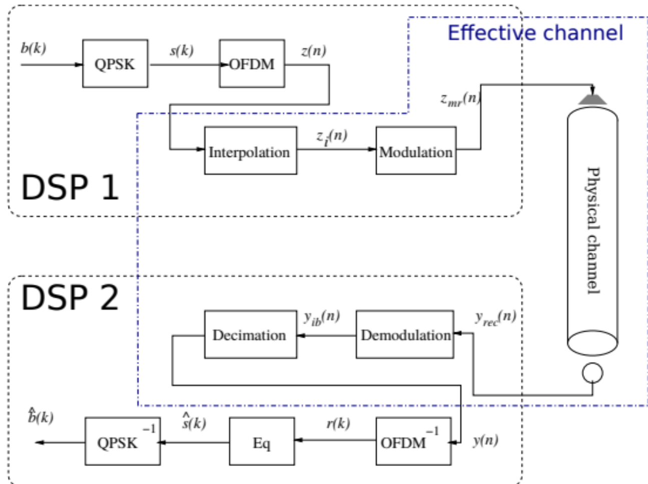

FIG. 1: Block diagram from project specification

II. Matlab implementation

1 Assume the baseband signal is interpolated by a factor of 8, a modulation frequency 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?

The final sampling frequency is 16 kHz, and the up-sampling factor is 8. So the bandwidth of the signal equals to $16/8 = 2kHz$. The up-sampling data will be periodic with frequency 16kHz and applying a filter with pass bandwidth of 2kHz, which results in a signal with approximately 2kHz and centered in zero. (magnitude between -1 and 1) Moreover, because the modulation frequency is 4 kHz, the signal will have the center frequency at 4 kHz. Therefore, the transmission band lies between 3 kHz and 5 kHz.

2 In part A we could achieve an EVM which was nearly zero if we set $snr = inf$. However, if we try to do the same thing with our new setup, we always get a nonzero EVM. Why is this the case? Note: this is not due to any non-ideality of the channel, as if we modify $simulate\_audio\_channel.m$ and add $h = 1$ to replace the simulated audio channel (after $h = impz(b, a);$) we will still see the same behavior.

The effect of nonzero EVM is due to the non-ideality of the low-pass filter that is used in interpolating. Due to the non-ideality, some noise (i.e. high magnitude signal) is beyond our filter bandwidth, which is mainly located in the transition band. Therefore, these signals cause distortion and aliasing, when we down-sampling and reconstruct the original signal.

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 the channel H? How do they contribute?

Theoretically, the effect of H will not appear in the modulation and demodulation part, since the demodulation is the sign-reversed carrier based on modulation, the effect can be fully removed.

Both taking the real part and propagation over the physical channel may contribute to channel H. Originally, our signal was located in the complex domain, taking the real part and discarding the imaginary part of the signal, we lose a part of magnitude that will be compensate later in channel H(equalization phase). The propagation over a physical channel affects the signal in phase and magnitude, such as attenuation and distortion. In addition, the lowpass filter is not ideal in the interpolation and decimation step, which may also introduce some effects of the channel due to the transition band limit.

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 the imaginary part of the signal. ***Hint: if z is the signal we wish to transmit, then discarding the imaginary part gives $z_r = \frac{1}{2}(z + \overline{z})$, where $\overline{z}$ is the complex conjugate of z. Furthermore, due to the linearity of the Fourier transform, we are also ensured that $Z_r(\omega)) = \frac{1}{2}(Z(\omega) + \overline{Z}(-\omega))$***

After interpolating and we modulate: $z_m(n) = z_i(n)e^{j2\pi n f_m/f_s}$

$Re\{z_m\} = z_{mr} = \frac{1}{2}(z_m + \bar{z}_m)$

$Z_{mr}(\omega) = \frac{1}{2}[Z_i(\omega - \omega_s \frac{f_m}{f_s}) + \bar{Z}_i(\omega - \omega_s \frac{f_m}{f_s})]$

according to the relation: $\bar{Z}(\omega) = Z(-\omega)$

$Z_{mr}(\omega) = \frac{1}{2}[Z_i(\omega - \omega_s \frac{f_m}{f_s}) + Z_i(-\omega + \omega_s \frac{f_m}{f_s})]$

Since in this case our channel can only carry a real-valued signal, and we can safely discard the imaginary part, while taking the real part of the signal does not lead to information lost, only add a mirror copy around ω_s (Nyquist frequency).

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 band width
– Phase linearity

In both stages, passband ripples and phase linearity are not pretty important. Still, it affects part of the performance, the passband ripple will affect the quality of the reconstructed signal in the interpolation stage, and excessive ripple will cause unwanted distortion. In the decimation stage, since the goal is to reduce the sample rate, while passband ripple is not that critical. While maintaining the phase linearity can be beneficial in preserving phase characteristics, both stages focus on handling the amplitude and sampling rate instead of phase accuracy, so it's not the most important factor.

In the system, in order to have a lower EVM value, we need to higher the stop-band attenuation or lower the transition bandwidth in lowpass filter in order to prevent aliasing and distortion.

If the transition bandwidth is too wide, the attenuation may cause ISI, and include other unimportant information. For the stopband attenuation, if it's not high enough, the lowpass filter cannot work as we desired and cause distortion and aliasing.

III. DSP-kit implementation

Let T, H, and R be the Fourier transforms of the transmitted signal, channel, and received signal respectively. The notation $\overline{x}$ will be used to indicate the complex conjugate of x.

1 The channel estimate is generated as $\hat{H} = \overline{T} \cdot R$. We avoid using the division operation because it is computationally much more demanding than performing a complex conjugation and multiplication. (Most computers generally perform some type of long division similar to the kind you learned in grade school, while multiplication is performed in one clock cycle). Assuming an ideal system, show that $\hat{H}$ and H have the same phase and magnitude.

Firstly, $H = \frac{R}{T}$ where $|H| = |R|/|T|$ and $\angle H = \angle R - \angle T$

Compared with $\hat{H} = \overline{T} \cdot R$ where $|\hat{H}| = |\overline{T}| \cdot |R|$

For complex number, $\angle \bar{T} = -\angle T$ and $|\bar{T}| = |T|$ so we can see $\angle H = \angle \hat{H}$ and $|\hat{H}| = |H||T|^2$

Because we use the QPSK which have the same transmitted magnitude that equals to 1, e.g. $|T| = 1$.

Therefore, $|\hat{H}| = |H|$.

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

By $R_{\text{eq}} = R \cdot \overline{\hat{H}}$, we can know that
$\angle R_{\text{eq}} = \angle R + \angle\overline{\hat{H}}$ and $\angle\overline{\hat{H}} = -\angle\hat{H} = -(\angle R + \angle\overline{T}) = -(\angle R - \angle T)$
Therefore, $R_{\text{eq}} = \angle T$
which shows that the angle of the equalized signal is equal to the transmitted signal.

3 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 does the EVM and constellation diagram change? Why?

When we operating with high signal amplitudes, we will get very small EVM that cause near-zero BER.

Then we low down the signal amplitudes, and make $SNR = 10dB$, we can see the constellation as FIG. 3. We have the constellation that spread a lot and cause large EVM and BER.

Getting signal constellation by using DSP, shown as FIG.4 and FIG.5, we get the $EVM = 1.233567$. We have assumed pilot: 'basic fishing_____' and assumed message: 'large heart_________'. We got the message: '_e_2_____c_R5P'

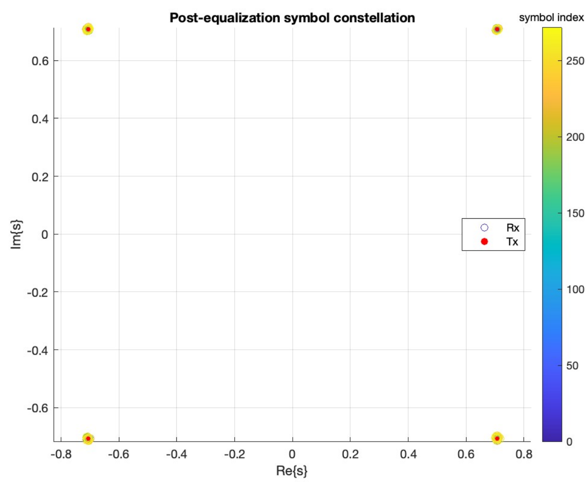

FIG. 2: Post-equalization constellation for high signal amplitudes using Matlab

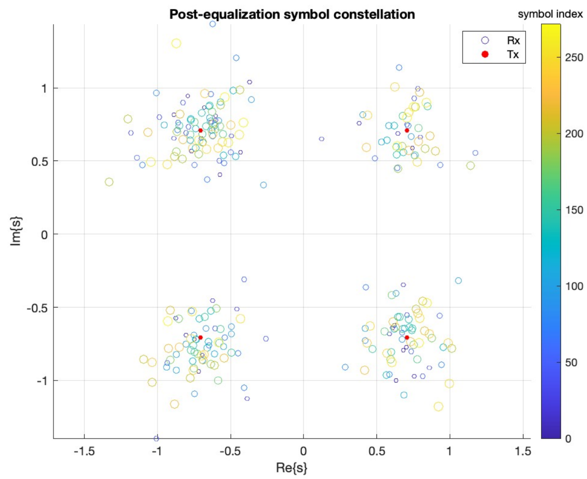

FIG. 3: Constellation for SNR = 10 dB using Matlab

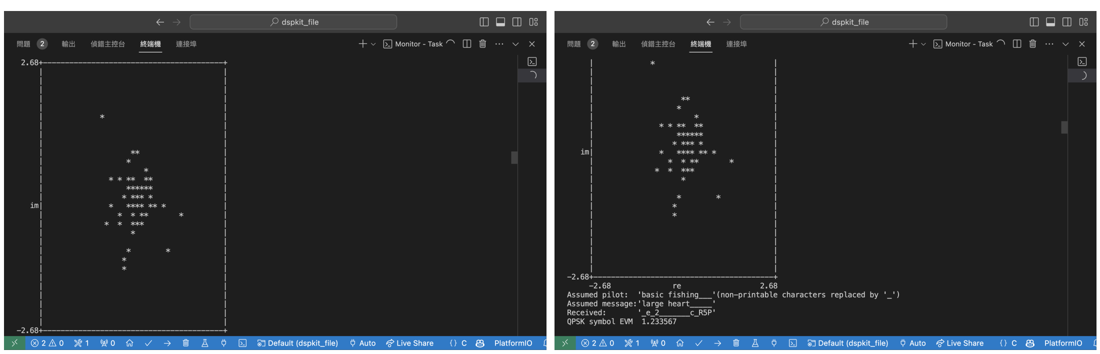

FIG. 4: Constellation for SNR = 10 dB using DSP and the shown message