UTKU ACAR 250206062

LAB-5 REPORT

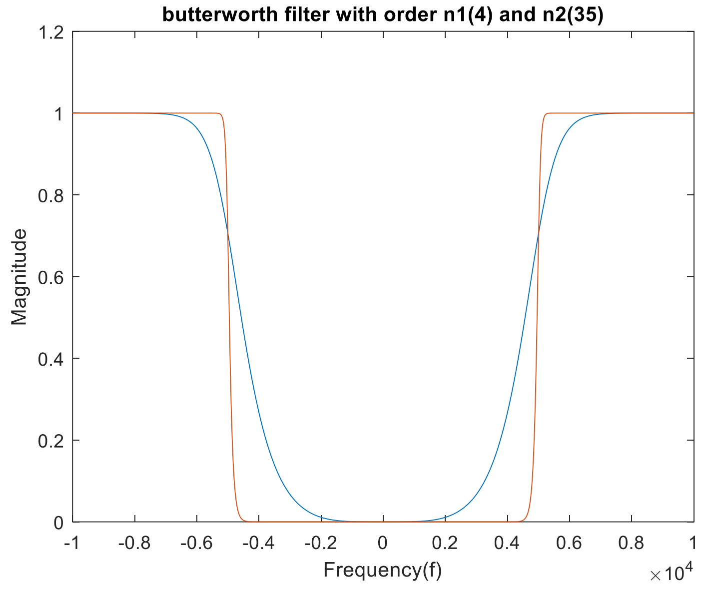

Figure 1: 2.1.d

In figure 1 we can see that as we increase the order of the filter the slope of the magnitude response at cutoff frequency also increases (gets more sharper) and passband magnitude is 1 as expected. n1 is blue n2 is orange.

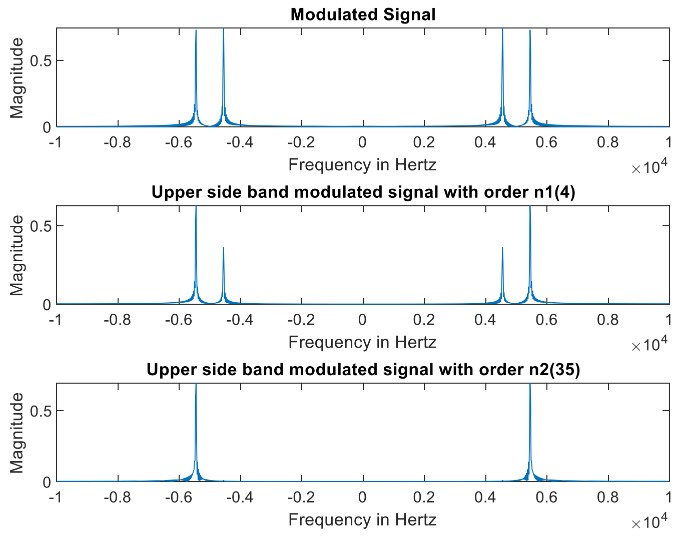

Figure 2: 2.1.f

In figure 2 we can see that we have 4 component at fc+fm (5450 Hz) and fc-fm (4550 Hz) and exact same at the negative side of the frequency vector(-fc-fm,-fc+fm) which is equal to (-5450Hz,-4550Hz). For the magnitudes they should be 0.75 because of trigonometric identities and Fourier transform halves the amplitude so our amplitude was 3 at beginning in time domain and we halved twice so we should get 0.75 at the output and MATLAB gives 0.72 and 0.74 as expected from non-idealities of the filters. The second figure shows us the insufficiency of the order of the filter to get the upper side band part of the signal. It does not suppress the lower side band enough and we can still see the lower side band even after filtering operation and has wrong magnitudes. The third one is the best order for our purpose to get upper side band while suppressing lower band sufficiently and keeping the upper side band unchanged as possible (0.695 can be accepted comparing with 0.74). I choose the susb2(t) signal because the order is sufficient to get nearly just the upper band part. Cutoff of the high pass filter is fc. The reason behind choosing high pass filter is we need to get the upper band signal.

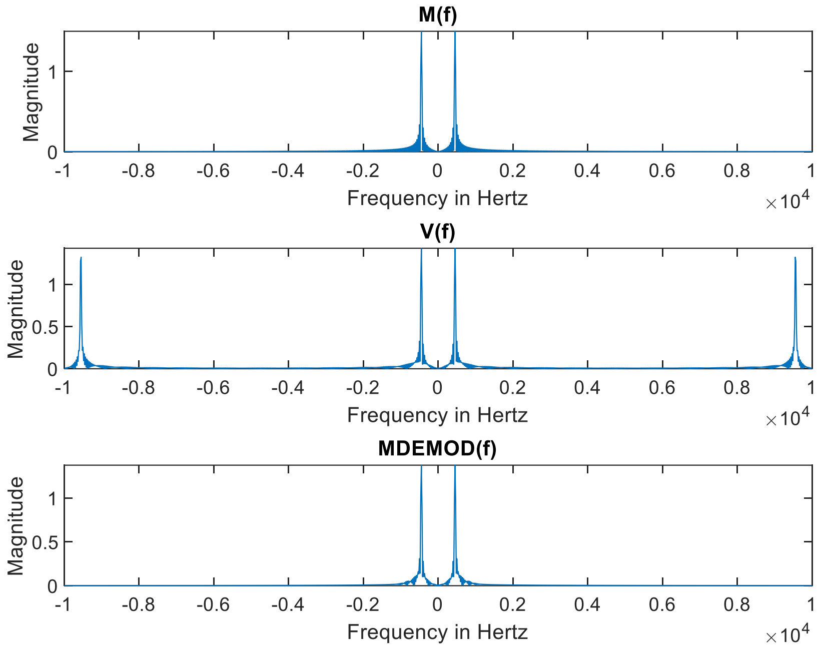

Figure 3: 2.2.d

I designed lowpass filter with order 9 and cutoff 2fm+20 for getting our message signal back. In figure 3 we can see that in first part our message signal has 1.5 magnitude which is expected because of the Fourier transform halves the amplitude and we get two peaks at -fm,fm as result with 1.5 magnitude. For second part we have 4 components at -2fc+fm, -fm, fm , 2fc+fm which can be explained by trigonometric identities again we multiplied modified carrier wave with 4 as amplitude and 0 degree phase difference with modulated wave s(t) and get v(t). I choose the amplitude as 4 due to this formula “Vo=Ac*Ac’*Am/4” from coherent detector formula for SSBSC which leads us to 3(our message signal amplitude) should equal to 1*Ac’*3/4 then we can see that Ac’ should be 4 to satisfy this equation. The 4 at denominator comes from using the trigonometric identities two times. First at modulation and second demodulation product modulators. We can see that magnitudes of the v(f) should be 1.5 and its nearly 1.5 (1.42). For third part after the low pass filter, we get rid of high frequency components and get only components at around our message frequency. For magnitude of the demodulated signal, we expect it should be same with our message signal (1.42) but we get 1.375 and this value can be accepted as long as they are close to each other.

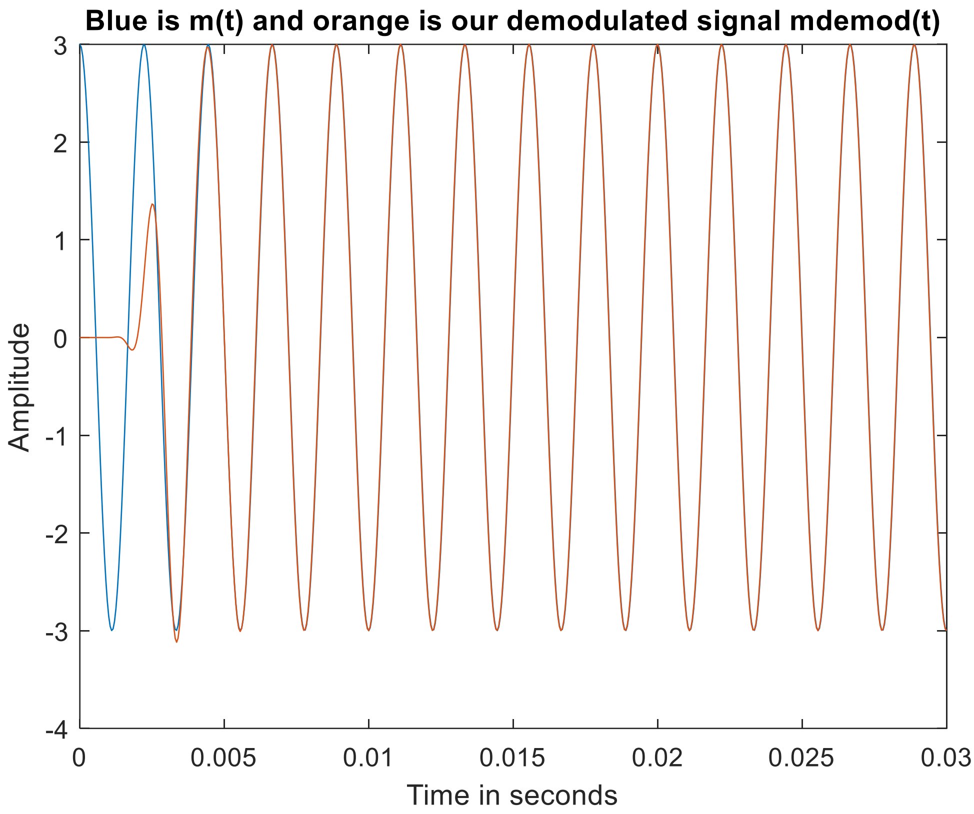

Figure 4: 2.2.e

As we can see in figure 4 the signals are more likely the same signal except first cycles. There are 0.02 amplitude difference (negligible) and 1 sample difference for same amplitudes. Demodulated signal is just 1 sample lagging our message signal which is also negligible.