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Fall

**Amplitude Modulation**

**Project**

2022

Communication

Name Sec BN

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**Explanation of our work:**

Starting off by reading our 3 input signals from the “Input” folder, we make sure that the sampling frequency in all these signals is the same for consistent modulation. Then, we choose the first channel for our messages and find the length of each one along with the maximum length. In order to be able to sum all modulated signals of each input, we need to make sure that they all have the same length so the next crucial step is to set their lengths to this maximum length and fill the gaps with zeros. We also adjust the time and frequency intervals on this max length and the original sampling frequency.

***Carrier Frequency:***

The carrier frequency is set to be the width of the signal with the largest width to ensure that when the signals are shifted during modulation, they do not overlap and fail to restore the input message signals. As for the second carrier frequency, it has to satisfy a condition where it covers the other signals by 2 \* Bandwidth plus another half of the width which is the same carrier frequency of the first. As a conclusion, we have two frequencies to work with: Fc\_1 which is the half width of the message signal drawn in the frequency domain and Fc\_2 which is triple that width.

***Modulation:***

We perform Double-Sideband Suppressed Carrier (DSB-SC) modulation on the first input signal with a cosine carrier of frequency Fc\_1. As for the second and third input signals, we use QAM (Quadrature amplitude modulation) where two signals are sent simultaneously over the same bandwidth (on the same carrier frequency Fc\_2) for efficient use of the bandwidth, but one is a sine function and the other is cosine. The difference in phase between these 2 functions helps us restore them separately without having to worry about the overlap that could happen.

***What about the final modulated signal?***

Now this should be the easiest step as all we have to do is sum the three modulated signals obtained from the DSB-SC and QAM to get our resulting modulated signal, ready for being used for demodulation.

***Demodulation:***

Our passband frequency for the low pass filter is the same as the original carrier frequency as all we need to do is filter each signal alone and output it for testing. For the synchronous demodulation with zero phase shift and frequency addition, the LPF takes the final modulated signal, the sampling frequency and the passband frequency. The signal is multiplied by 3 carriers, one for each input signal, obtaining an output of half the original signal in amplitude so we simply finish by multiplying by 2 and passing the necessary parameters mentioned above to the low pass filter to get our audio back again.

***Demodulation with phase shift:***

The same exact steps done with the synchronous demodulation will be done here but the only difference is that we add a phase shift with the carrier inside the cosine function. As required, we have a phase shift of 10, 30, and 90 degrees (all converted to radians by a simple calculation of course) which will affect the output signal. Phase shift causes attenuation of the signal which means decreasing its amplitude. However, there is a special case for phase shift 90 because it causes the first signal to be almost gone and can’t be heard as cosine (90) is zero. For the other 2 signals, we used QAM by applying cosine and sine for the same frequency and when adding a 90 degree phase shift, the two signals interchange. One takes place of the other and vice versa because 90 phase shift causes cosine to become sine and the opposite as well.

***Demodulation with a different local carrier frequency:***

A constant is added with the carrier frequency that is calculated with Omega and then we observe the effect of that on the carrier when it’s multiplied by the modulated signal. For this, we have 2 cases where we add this constant on Fc: 2Hz and 10 Hz. The effect adding a 2Hz constant is distortion of the signal and a bit of noise will be clearly heard in the background but we would still be able to identify the audio from the main core of the signal as the effect wasn’t that much drastic. On the other hand, with 10 Hz the signal will be mostly disrupted by random noises and we would fail to know which audio is currently playing.