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ECE-3323-002
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Modulation Project

DSB Modulation and Demodulation

Double Sideband modulation and demodulation is the easiest type exhibited in this project. The premise is, to increase signal frequency to a level fit for long-distance travel, the original signal, $m(t)$, is multiplied with a carrier signal, $\cos(2\pi f_c t)$, where f_c is the carrier frequency. The carrier frequency in this case is 2 kHz and will remain so for all subsequent exhibits in the project. As seen in Figure 1, this process simply shifts the original signal to be centered around a higher frequency. Figure 2 shows a detailed view of how the process affects the waveform. In essence, the profile of the original signal is upheld, allowing for a simple demodulation.

The demodulation step is simply multiplying the new signal $s_{DSB}(t)$ with $2\cos(2\pi f_c t)$, and then filtering the excess signals out with a low-pass filter, causing the final recovered signal $s_{REC}(t)$. The multiplication essentially regenerates the original signal in its original place since it is doing a shifting process again. However, it also generates some high-frequency signals, which is what the low-pass filter removes. In our case, we set the ideal low-pass filter to have a 500 Hz bandwidth. This is far lower than initially expected, though this can be explained by the linear ramp involved in the filter thus allowing frequencies above desired to be let through. This can be proven by looking closely at the final signal waveform (not pictured here). At any bandwidth above 500 Hz, $s_{REC}(t)$ is jagged from artifacts of the carrier frequency. However, at 500 Hz, the signal matches exactly with $m(t)$.

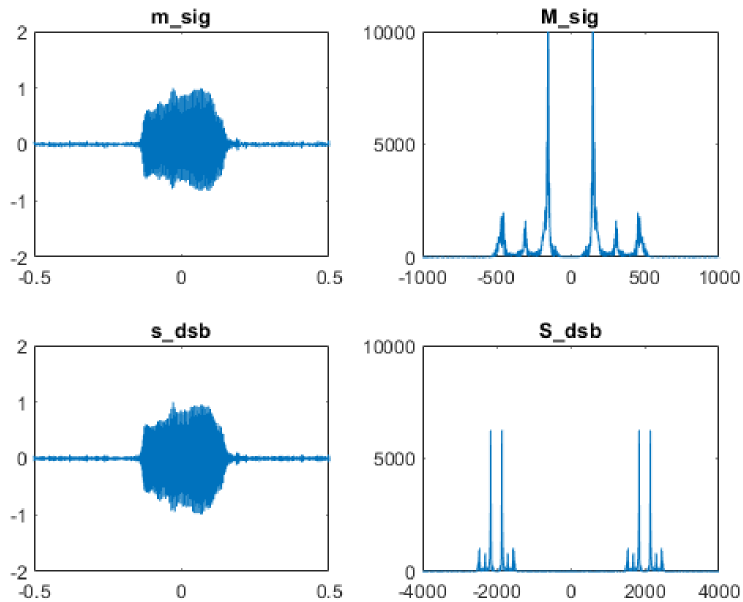

Figure 1: DSB modulation. The top two figures are the original signal and the bottom two the modulated, then from left to right for each is the signal in the time domain and the frequency domain.

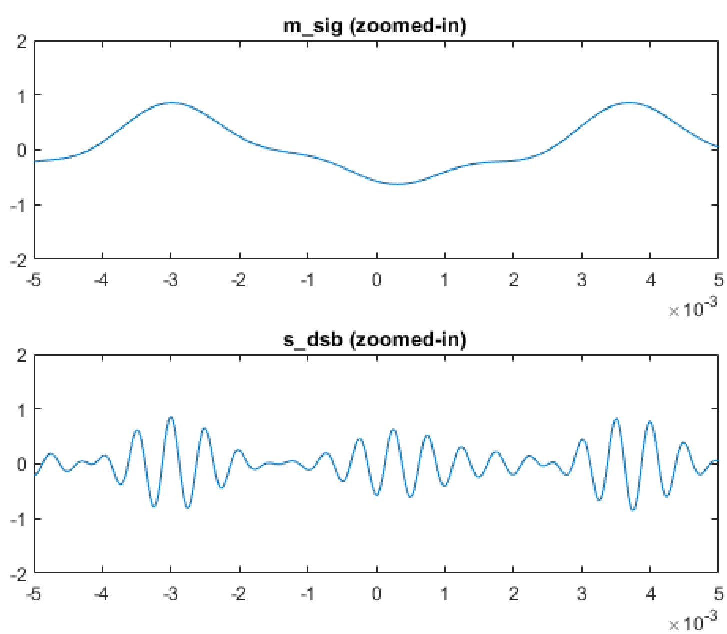

Figure 2: A closer look at DSB modulation in the time domain.

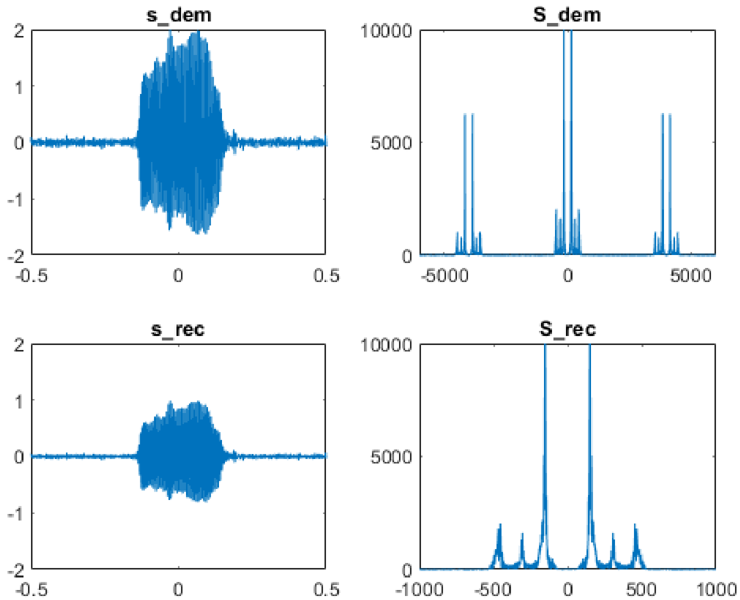

Figure 3: Demodulation of the message signal. On top, the initial demodulation stage, below is the final signal after the low-pass filter.

AM Modulation and Demodulation

Amplitude Modulation is the next step in complexity for fitting signals in a transmittable format. AM involves multiplying *and* adding $m(t)$ with the carrier signal $\cos(2\pi f_c t)$. This creates a DC offset that effectively makes the frequency of the transmitted signal $s_{AM}(t)$ steady, thus encoding $m(t)$ purely through amplitude. One may notice that the subsequent signal, seen in Figure 4, remains loud during the normally quiet portion of the original waveform. This is because there is still a small amount of background noise present, and $s_{AM}(t)$ encodes it as a medium amplitude.

To demodulate $s_{AM}(t)$, we use a rectifier on the incoming signal, which essentially outputs just the magnitude of the amplitude of the signal. This produces a very noisy signal, $s_{DEM}(t)$ and retains the aforementioned DC offset. Both elements are visible in the spectral graph in Figure 5. To clean this noise up, we need to employ both a low-pass and a high-pass filter. The low-pass filter is tuned more aggressively, at 350 Hz since the noise is present far closer with the desired information. The high-pass filter is primarily used to remove the DC offset, but it also does some cleaning-up. It is an option to simply use a DC-blocking filter, however it will not clean up the low frequencies like a high-pass filter will, and since this project focuses on speech, we do not particularly worry too much about losing low-end that did not particularly exist to begin with. Figure 6 shows the result, using a high-pass filter. The high-pass filter is biased at having a bandwidth of 150 Hz. Note that the $s_{RECBLK}(t)$ is shifted down about 0 compared to $s_{REC}(t)$ and that the matching spectral graph $S_{RECBLK}(t)$ does not possess the enormous spike at 0 Hz. Also noticeable is that the $s_{RECBLK}(t)$ is not as smooth as the $s_{REC}(t)$ of the DSB modulation scheme. That is because AM inherently produces noise even among the frequencies of the original signal, unlike DSB, which creates a clear separation between the original signal and artifacts, as can be compared between Figures 3 and 5.

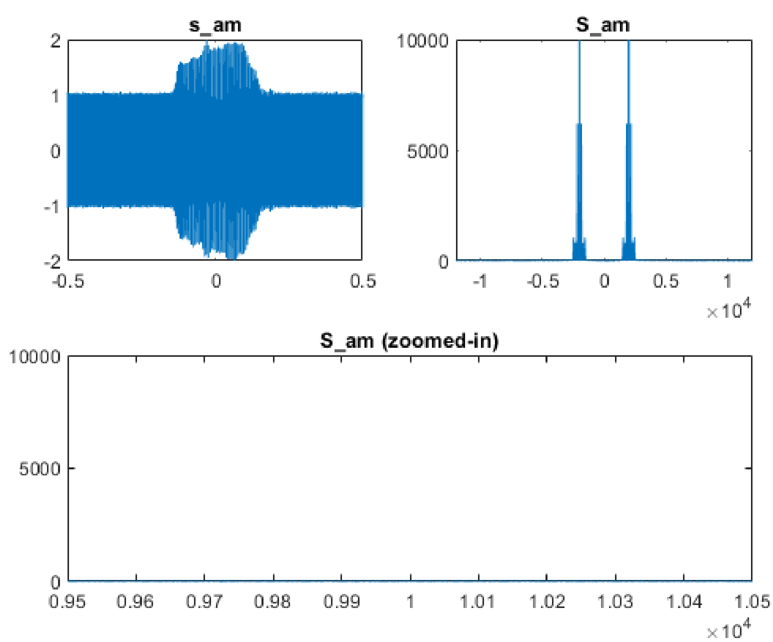

Figure 4: AM of $m(t)$

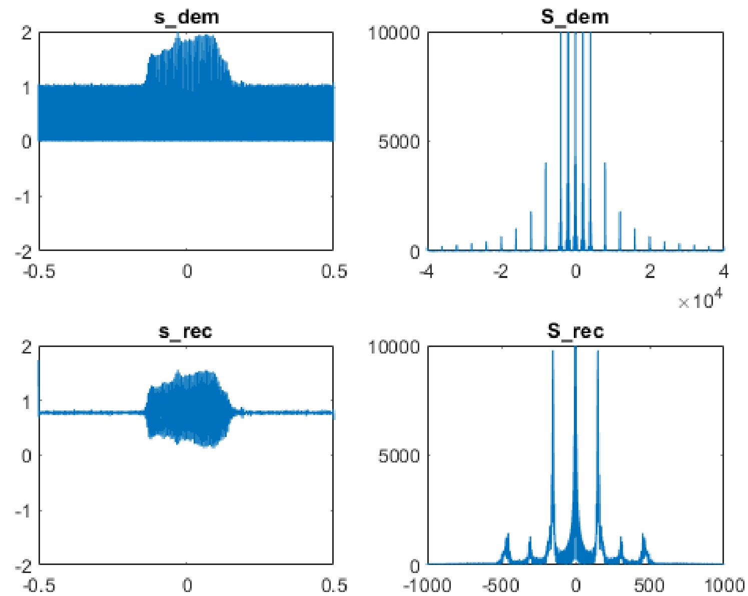

Figure 5: AM demodulation and application of low-pass filter.

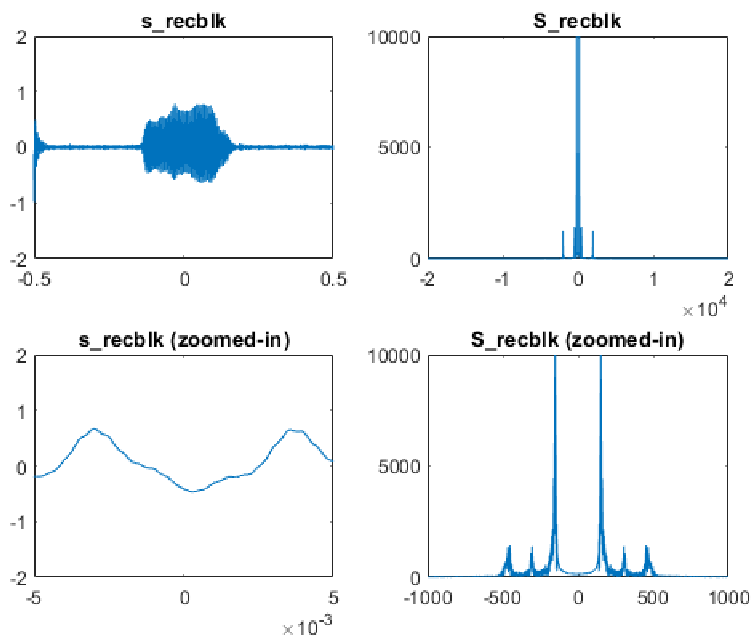

Figure 6: Application of high-pass filter on AM demodulated signal

FM Modulation and Demodulation

The final scheme exhibited in this project, Frequency Modulation, increases the complexity by yet a couple more steps. $m(t)$ is first integrated and multiplied by the factor k_f. These two operations fix the signal at a specific amplitude. Then, the familiar $\cos(2\pi f_c t)$ is added with the integrated and multiplied $m(t)$ to produce $s_{FM}(t)$. This signal thus encodes all the information in terms of frequency and change of frequency. Comparing Figure 7 to 2, we can see that as $m(t)$'s amplitude increases, $s_{FM}(t)$'s frequency increases. Therefore, the frequencies of the original signal are simply based on how the frequency of $s_{FM}(t)$ changes. Since the amplitude remains consistent, the power to output the signal is $\frac{amplitude^2}{2}$, allowing for a very predictable system.

Demodulation in this situation follows the same idea of the prior two: reversing the steps done then filtering the artifacts. First, taking the derivative $s_{FM}(t)$ and dividing by k_f retrieves the amplitude information, as seen in Figure 8. Once a low-pass and high-pass filter is applied in the same way as the AM situation (however with a tweak in tuning the low-pass filter's bandwidth a bit lower at 285 Hz), then all that remains is the final recovered signal $s_{REC}(t)$. If we zoomed in on the waveform in Figure 9, there would be some distortion, however it is a better result than in the AM scheme and there is no discernable difference from a distant perspective, as seen in the near-perfect overlapping graph in Figure 9. Something to point out is that from the spectrum perspective, like the AM scheme, this still possesses some low-end noise from the DC component, imperceptible as it may be. One final note about both AM and FM schemes: after all the filtering was completed, we amplified the final signals by 2.5 so that they may be about equal in loudness with $m(t)$.

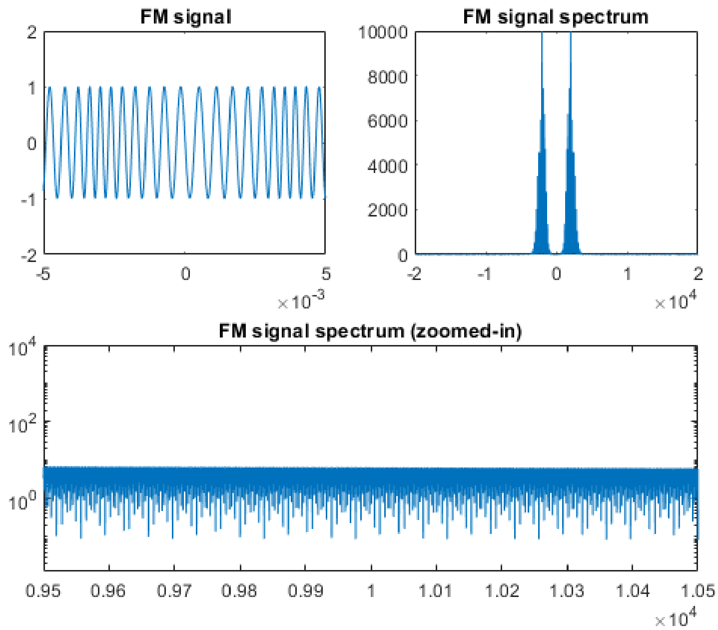

Figure 7: FM of $m(t)$

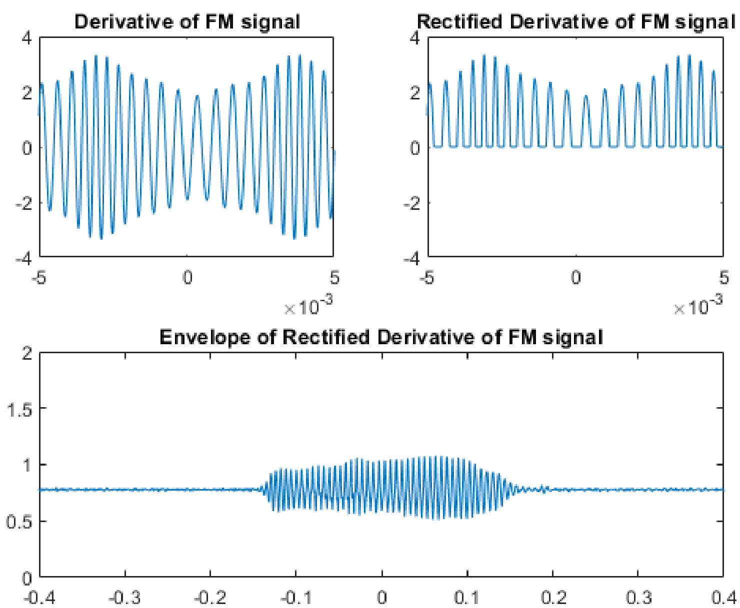

Figure 8: Derivative, Rectified Derivative, and Envelope of Rectified Derivative of $s_{FM}(t)$

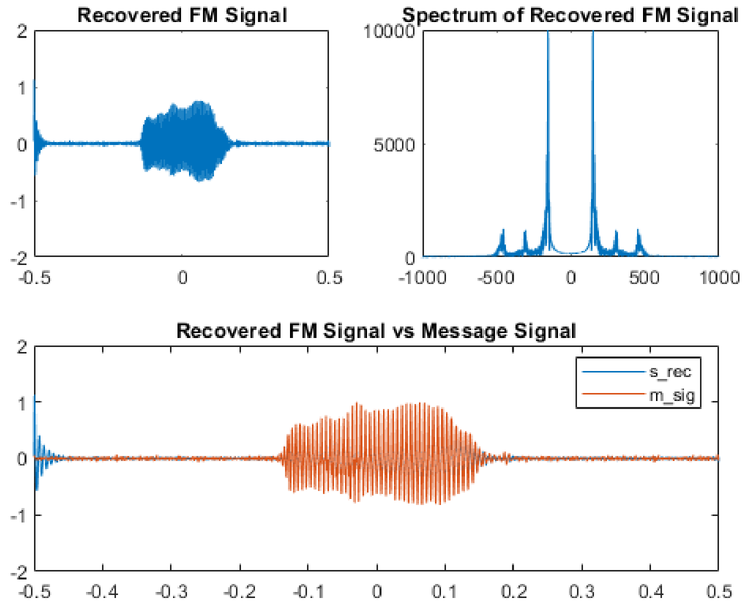

Figure 9: Recovered $m(t)$ from $s_{FM}(t)$. (Note: the initial wave at $t = -0.5$ in the bottom graph is due to mathematical process and is not present in a real system)