WES268A Fall 2015
Digital Communications
Lab 1: Prelab

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1 Theory Problems

1.1 Spectrum of AM modulated signals

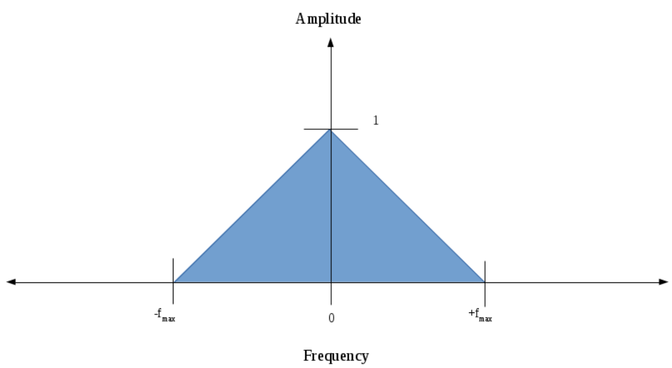

Figure 1: **A sketch of the frequency spectrum of the baseband signal $S(f)$.**

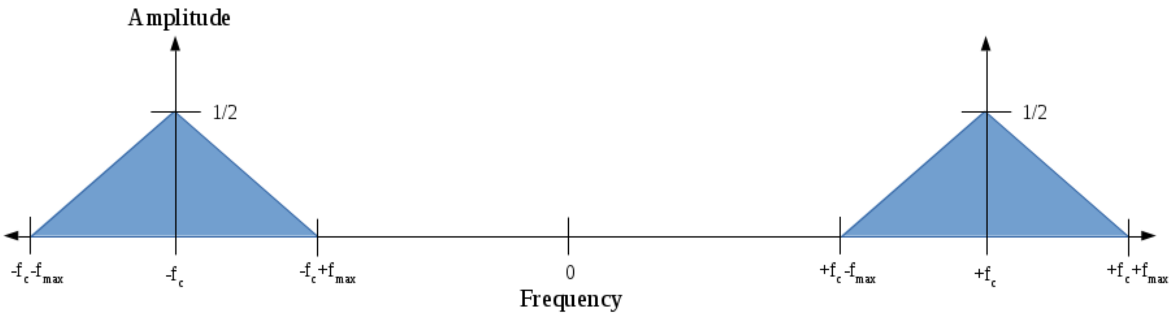

Figure 2: **A sketch of the frequency spectrum of the amplitude-modulated pass-band signal $\tilde{S}(f)$.**

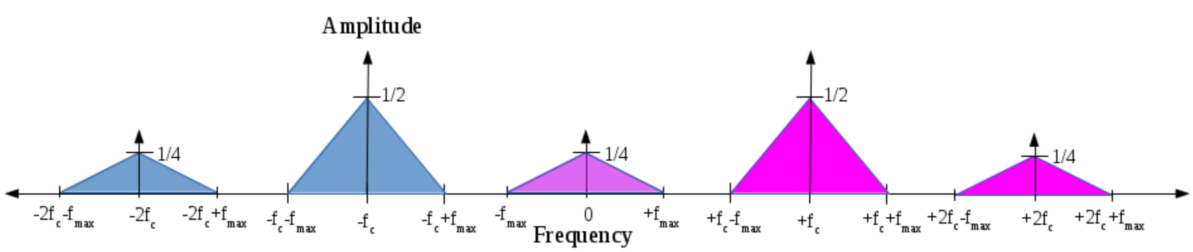

Figure 3: **A sketch of the frequency spectrum of the demodulated signal.**

1.2 Frequency demodulation errors

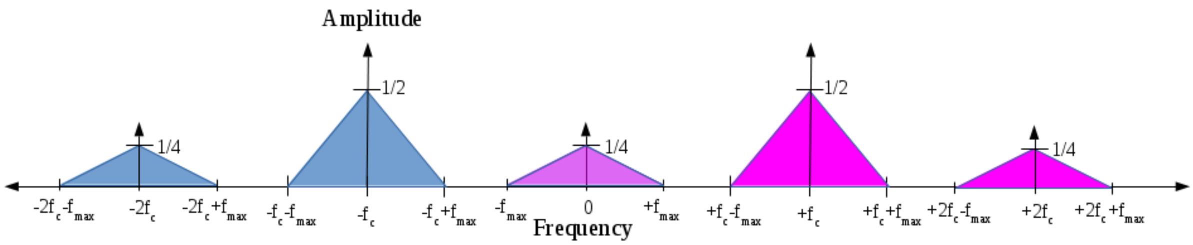

Figure 4: **A sketch of the frequency spectrum of the demodulated signal with frequency error $x = 0$.**

As frequency error is introduced, the aliased baseband copies shift past each other. Because f_c dominates f_{max}, we assume f_c is at least several orders of magnitude larger than f_{max}.

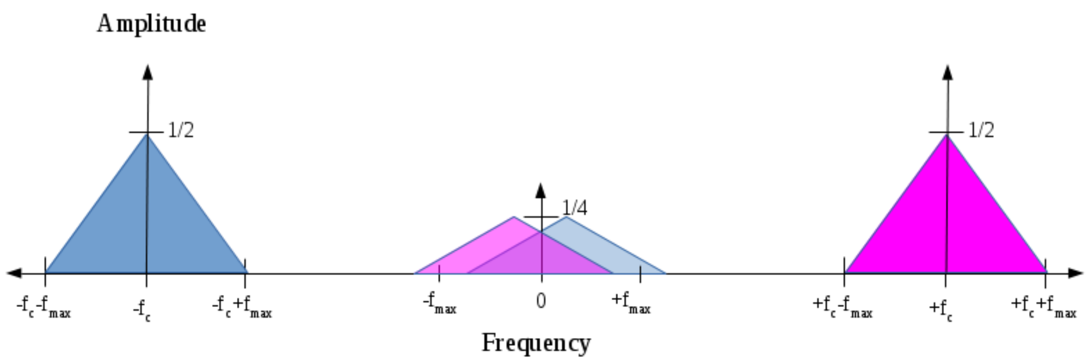

Figure 5: **A sketch of the frequency spectrum of the demodulated signal with frequency error $x = \frac{f_{max}}{10 f_c}$. Note that spectral copies at $\pm 2 f_c$ exist but are not visible in the figure.**

The frequency error is now a magnitude larger than before, but is still relatively small because f_c is several order of magnitude larger than f_{max}. Even as the aliased copies slide past each other at baseband with the introduction of frequency error, the Nyquist rate of the baseband signal is unaffected because f_{max} remains unchanged.

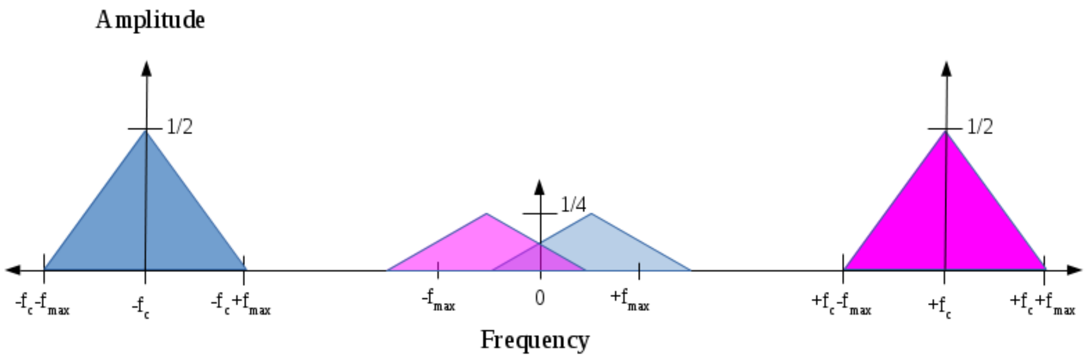

Figure 6: **A sketch of the frequency spectrum of the demodulated signal with frequency error $x = \frac{f_{max}}{f_c}$. Note that spectral copies at $\pm 2f_c$ exist but are not visible in the figure.**

1.3 Phase demodulation errors

An AM signal $\tilde{s}(t) = A\ cos(2\pi f_c t)$ where A is a constant is demodulated by $cos(2\pi f_c t + \phi)$ where ϕ represents a phase error.

An expression for the demodulated signal $d(t, \phi)$ as a function of the phase error ϕ is given by:

$$d(t, \phi) = A\ cos(2\pi f_c t) cos(2\pi f_c t + \phi) \tag{1}$$

The period of the carrier f_c is given by $T = \frac{1}{f_c}$.

If the demodulated signal $d(t, \phi)$ is integrated over a time period T that is many times the period of the carrier (i.e., $N\ T$, where $N >= 2$), the value of the integral without phase error ($\phi = 0$) is given by:

$$\begin{aligned} M_0 &= \int_0^{\frac{2}{f}} A\ cos(2\pi f_c t) cos(2\pi f_c t) \\ &= \frac{1}{f} \end{aligned} \tag{2}$$

The value of the integral with phase error ($\phi \neq 0$) over the same period is given by:

$$\begin{aligned} M_1 &= \int_0^{\frac{2}{f}} A\ cos(2\pi f_c t) cos(2\pi f_c t + \phi) \\ &= \frac{cos(\phi)}{f} \end{aligned} \tag{3}$$

The maximum phase error ϕ that can be tolerated for the demodulated signal to ensure the amplitude is within ten percent of the amplitude without a phase error is given by:

$$
\begin{aligned}
\frac{M_0}{M_1} &\leq 10 \\
sec(\phi) &\leq 10 \\
|\phi| \leq &\ sec^{-1}(10) \\
|\phi| \leq &\ 0.4706
\end{aligned}
\tag{4}
$$

1.4 Agilent tutorial on Spectrum Analyzers

The resolution bandwidth required to resolve signals separated by 50 kHz that differ by 40 dB in power is given by:

$$
-\ 40\ log_{10}\left(\left(\frac{50000}{\frac{B}{2\sqrt{\sqrt[4]{2}-1}}}\right)^2 - 1\right) = -40\ dB \tag{5}
$$

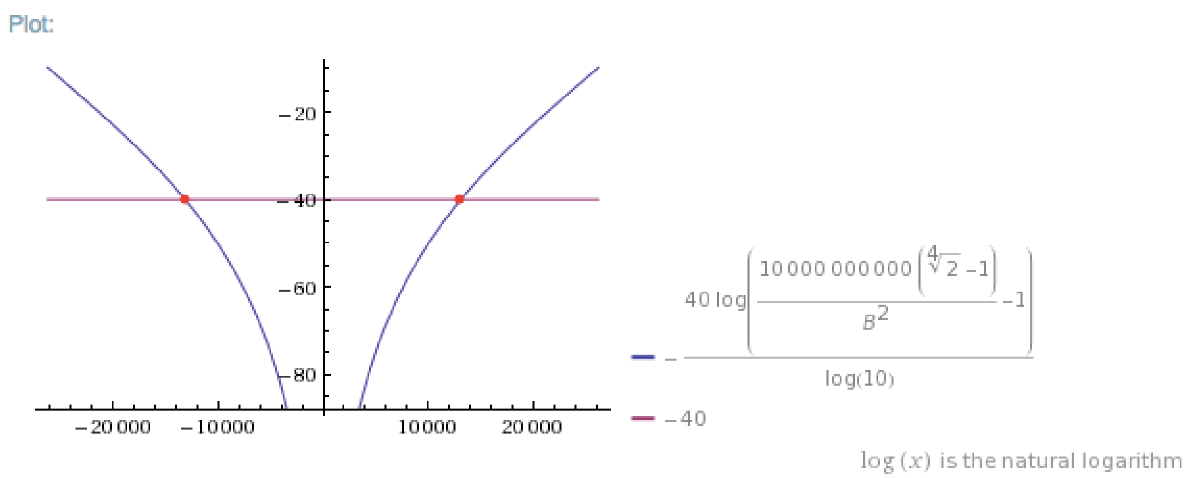

Figure 7: **Calculating the resolution bandwidth that meets design specifications.**

Solving for the resolution bandwidth $B \approx 13115\ Hz$.

Changing the video bandwidth (filter) or averaging will affect the result. If the video bandwidth is changed, the resolution bandwidth will also need to change. Averaging acts as low-frequency filter that rejects high frequency noise, effectively smoothing the signal. This reduction in noise will also affect the required resolution bandwidth.

The expected value of additive noise is zero averaging acts a low-pass filter that smooths the signal and eliminates the additive noise. This reduction in noise will affect the required resolution bandwidth.

2 Matlab/Simulink Simulations

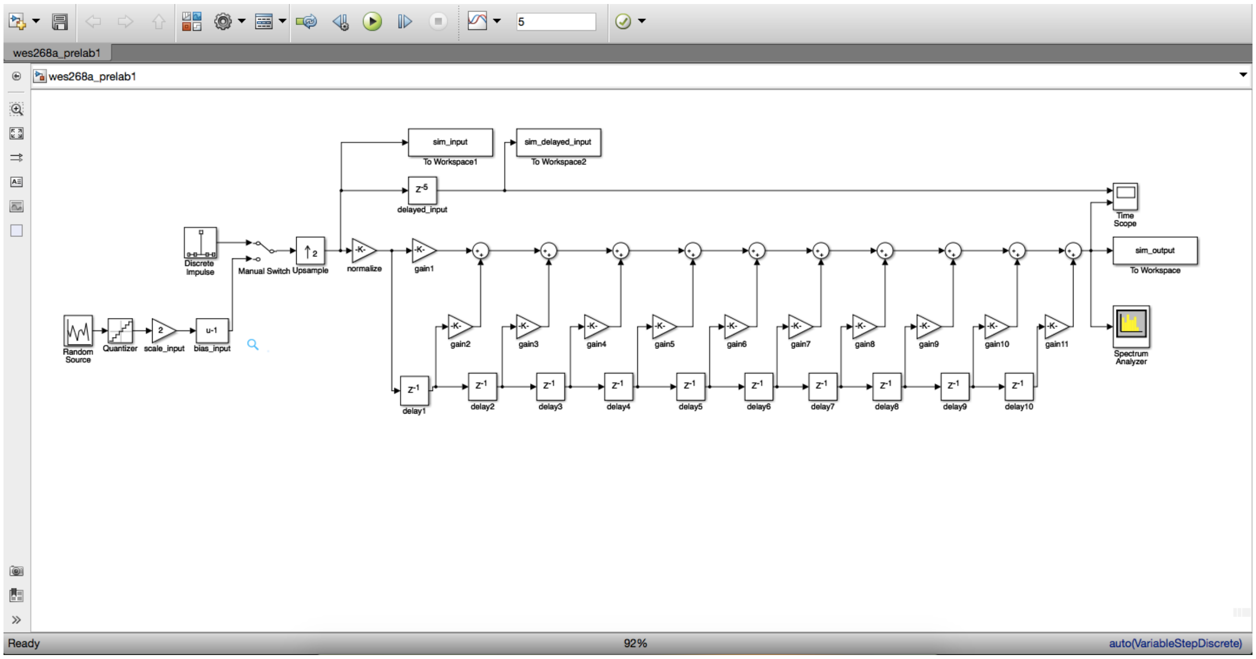

Figure 8: **A simulink model that implements a raised cosine pulse with** $\beta = 0.5$**,** $delay = 5\ samples$**, and** $rate = 2\ samples/symbol$**. A switch is provided to toggle between measuring the filter's impulse response and the filter response to modulated symbol data. An upsample block is provided to achieve the required sample rate.**

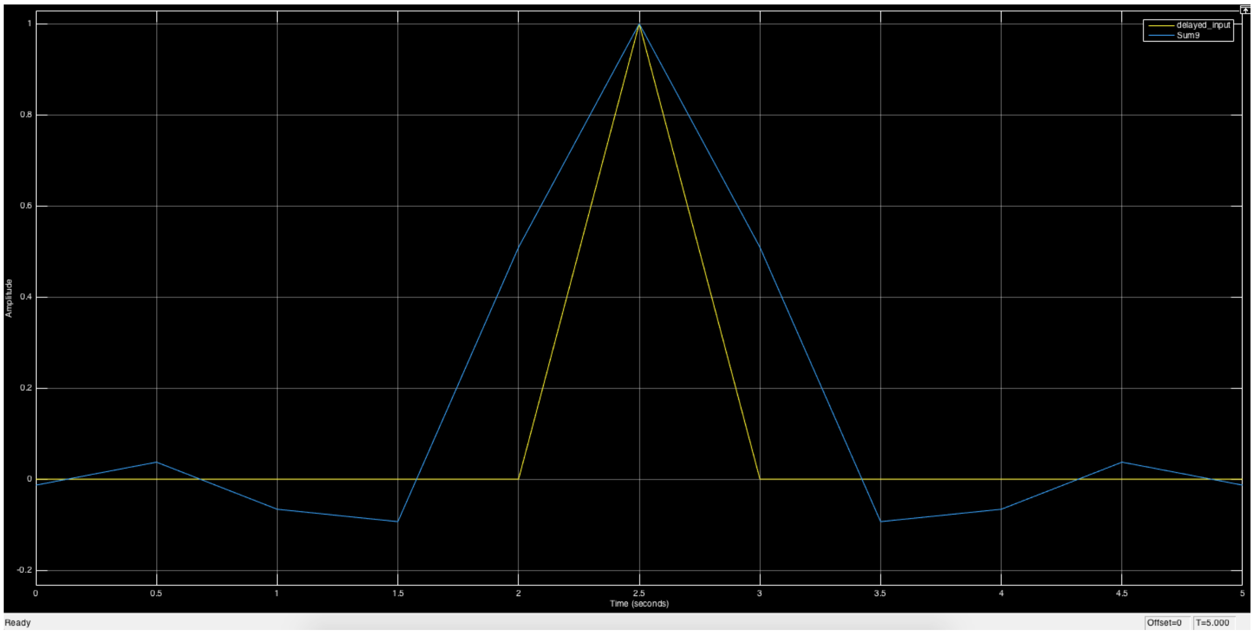

Figure 9: **The filter impulse response.**

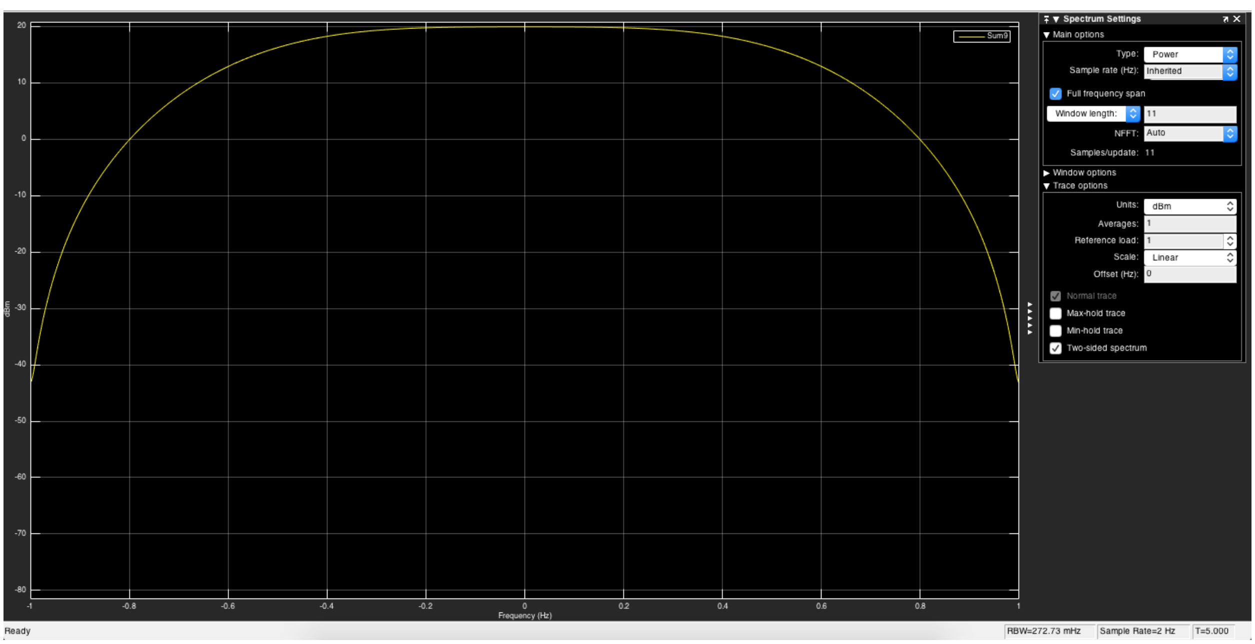

Figure 10: **The filter frequency response.**

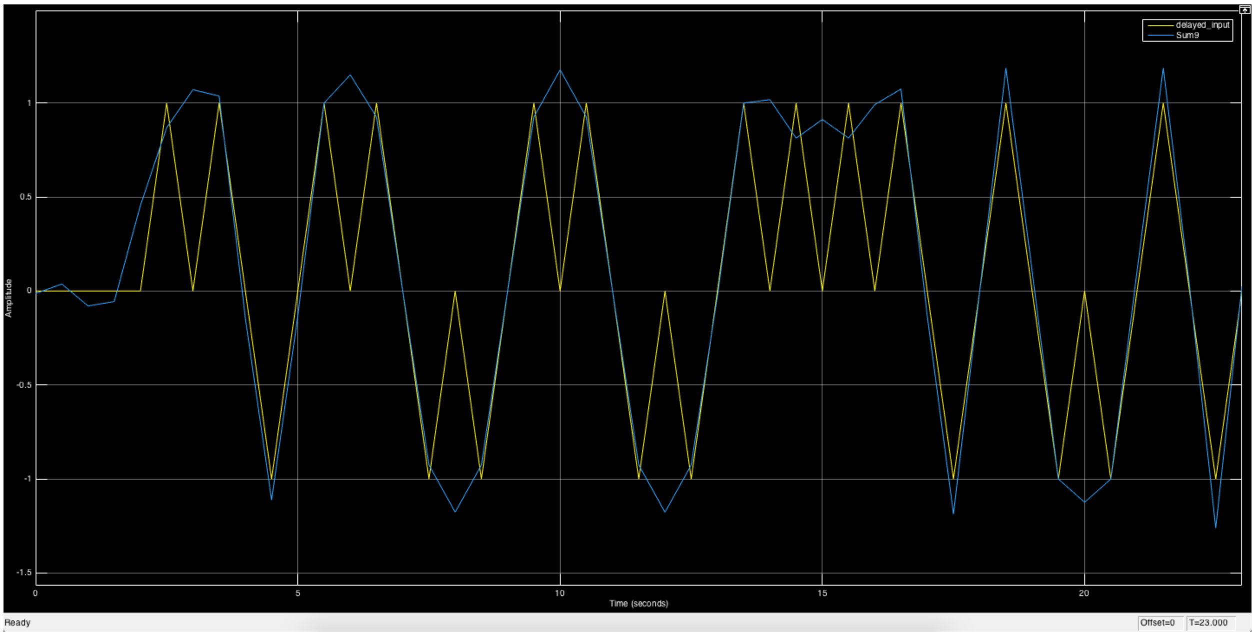

Figure 11: **The filter input and output for at least 20 symbols of upsampled data.**

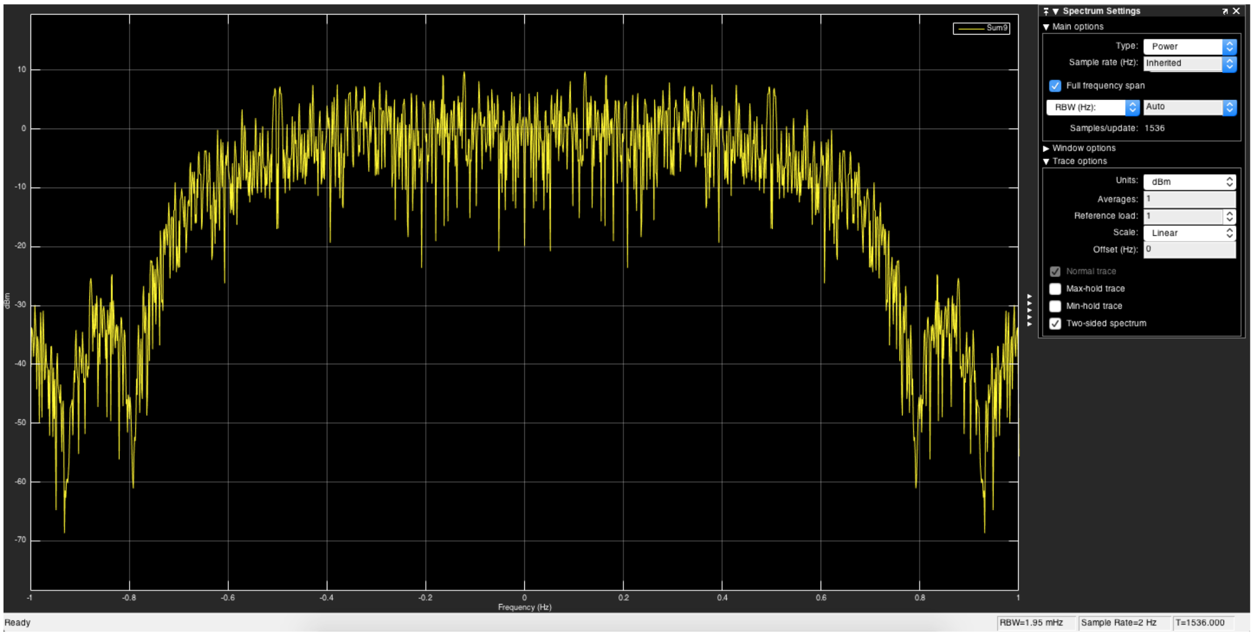

Figure 12: **The filter output spectrum for modulated BPSK data. The spectrum looks like a windowed function.**