Capsim Application Note

Analog Modulation and Demodulation

Introduction

Various analog modulation schemes can be simulated in Capsim using built in stars. In this application note, we will present three example simulations: Double Sideband Suppressed Carrier (DSBSC) with carrier recovery, Single Sideband (SSB) modulation/demodulation, and Frequency Modulation (FM). In SSB modulation we will use Hilbert transforms for modulation and demodulation. In the simualtion of analog FM modulation, we will show how Capsim simulations match mathematical predictions of the FM magnitude line spectra.

Double Sideband Suppressed Carrier (DSBSC) Modulation with Carrier Recovery

The modulation of a baseband signal using DSBSC is straight forward. However, to demodulate a DSBSC signal, we need to recover the carrier for synchronous demodulation. The toplevel block diagram of a DSBSC system is shown in Fig. 1. Information bits are generated by the *data* star. Each bit ("1" or "0") is encoded into a bipolar signal by the *coder* star. Thus, a one is mapped to +1 and a zero to -1. The *coder* also increases the sampling rate by 8 by inserting 7 zeroes between successive bipolar symbols. The pulse shaping star then produces a pulse stream with zero intersymbol interference using raised cosine Nyquist pulse shaping (with a rolloff factor of 100%). Let us assume that the bit rate is f_s. The sampling rate at the output of the pulse shaper is now $8*f_s$.

Usually, in DSBSC modulation, the carrier frequency is much larger than the baseband bandwidth. Thus, we must increase the sampling rate of the baseband signal by, say, a factor of 100. This is done by the *resample* star. This star has several parameters. One parameter determines the oversampling rate, 100 in this case, and another parameter sets the interpolation algorithm. Hence, at the input to the mixer, we have a continuous analog signal (it is actually a discrete sampled signal but the sampling rate is very high compared to its bandwidth).

The oscillator connected to the mixer generates a sine wave at a frequency of 1 kHz. The DSBSC signal is available at the mixer output. The spectrum of the DSBSC signal is shown in Fig. 2. Note the raised cosine shape of the spectrum. Also, note that there is no distinct carrier present.

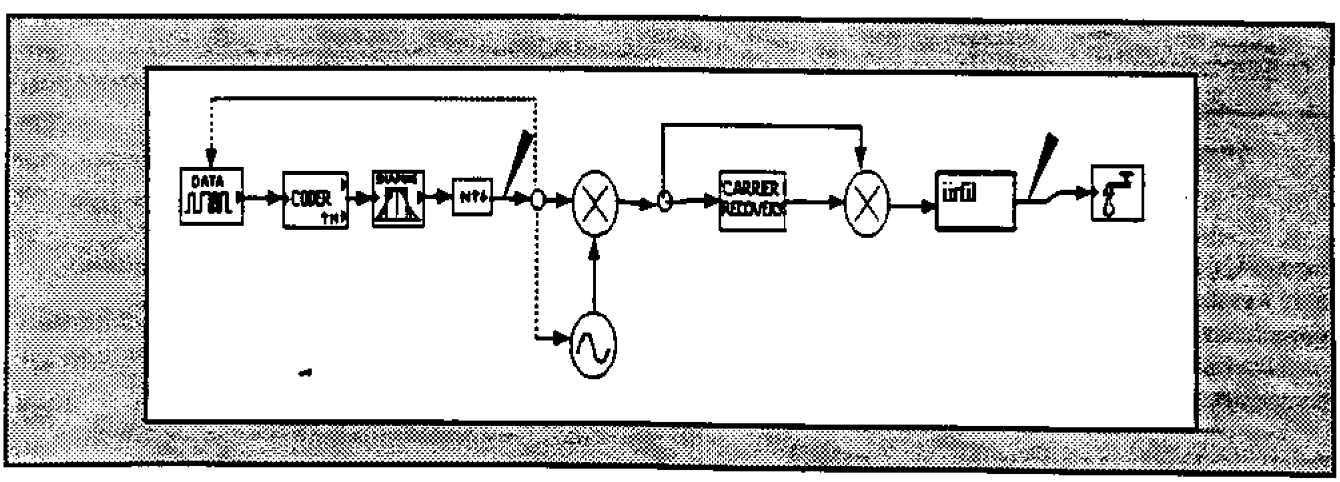

Figure 1. Toplevel topology for DSBSC modulation/demodulation

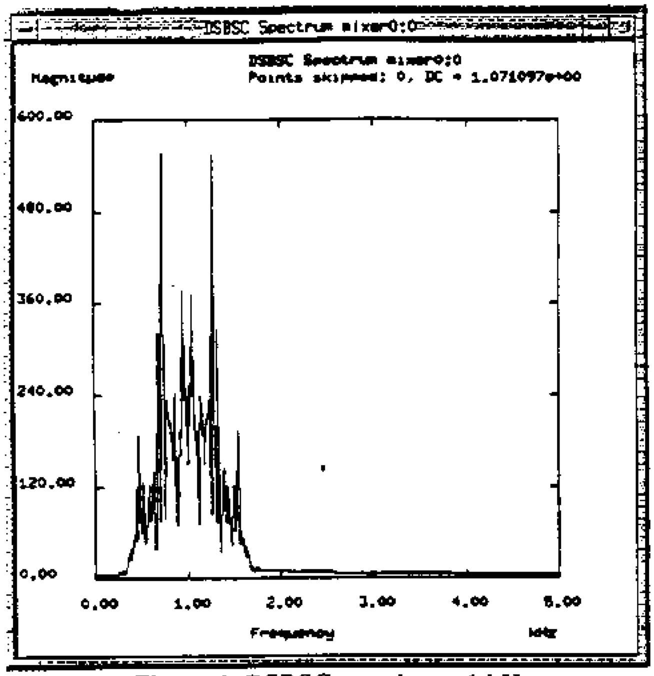

Figure 2. DSBSC, carrier at 1 kHz.

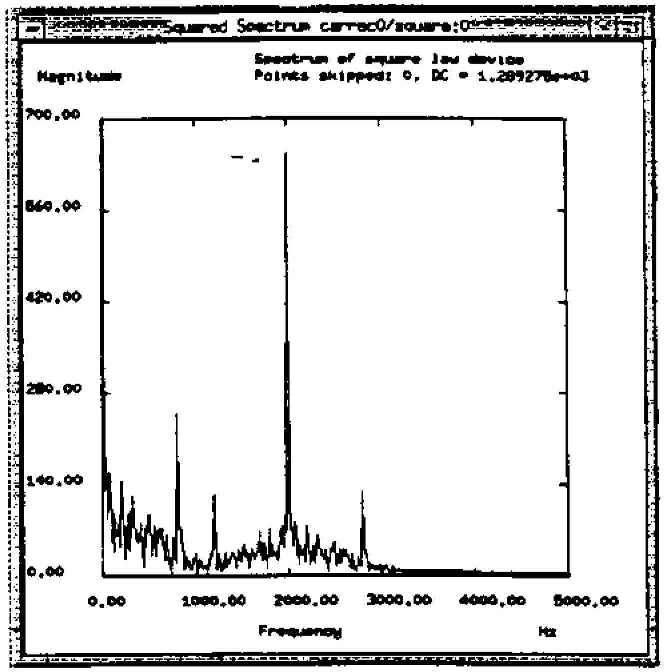

Figure 4. Spectrum at the output of the square law device.

Demodulation of DSBSC Signals

To demodulate the DSBSC signal, we must first generate the carrier from the received signal. This is done using a galaxy, *Carrier Recovery*. The output of the carrier recovery galaxy is mixed with the DSBSC signal. The baseband component is recovered by the low pass IIR filter (an elliptic filter) which suppresses the component at twice the carrier frequency. Fig. 3 shows the block diagram of the *Carrier Recovery* galaxy. The system consists of a square law device, a tuned bandpass filter (using the *bps* star) and a divide by two star (a flip flop). In an analog system, there is no need for the bandpass filter at the output of the flip flop. However, in a computer simulation of the system, the sampling rate is finite. If the square waveform is used in the mixer for demodulation, the spectral components of the harmonics of the fundamental will alias and

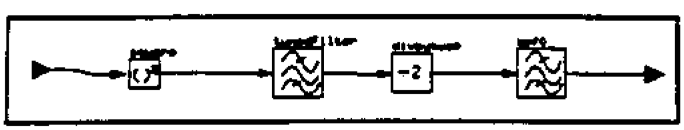

Figure 3. Carrier recovery galaxy

errors may result. Actually, a bandpass filter is not such a bad idea in a real system since the higher harmonics act as interference in the mixer. The spectrum at the output of the square law device is shown in Fig. 4. Note the large spectral component at 2 kHz (twice the carrier).

The demodulated baseband signal is plotted together with the original baseband signal in Fig. 4. Note that the

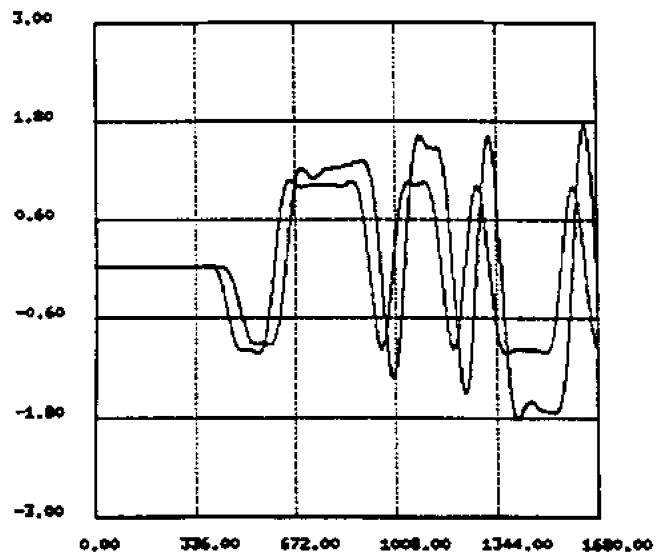

Figure 5. Baseband and demodulated waveforms.

recovered baseband signal is in very close agreement with the original baseband signal. An apparent time constant is also evident from the plot. The time constant is related to the Q of the tuned bandpass filter in the carrier recovery loop. By increasing the Q better noise immunity is achieved. The transient time, on the other hand, increases. Also, any mistuning may substantially reduce carrier power for high Q values. All these issues can be investigated by changing the appropriate parameters in the Capsim simulation of the DSBSC system.

FM Modulation

A simple FM modulator can be constructed using the dco star. This star is a digitally controlled oscillator. The dco star produces an in-phase and quadrature output signal with constant modules. The frequency, however, is directly proportion to the level at the dco input. Thus, the simple FM modulator shown in Fig. 11 is built from a sine wave generator, the sine star, and the dco star. The sink star is used to absorb the output samples of the dco.

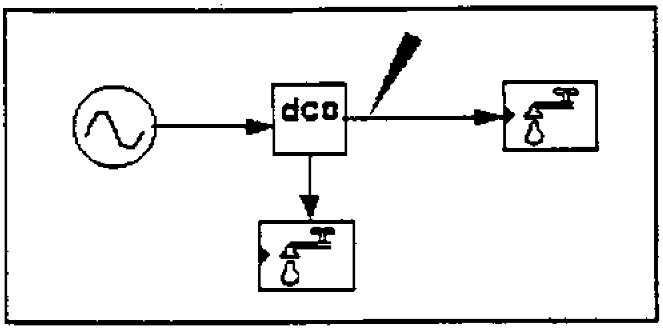

Figure 12. FM Modulation

The frequency at the dco output is related to the input level by the expression,

$$f = f_0 + v_{in} \frac{f_s}{2\pi}$$

where v_{in} is the input level, f_s is the sampling rate, and f_o is the dco center frequency.

To simulate an analog FM system, we select an appropriate sampling rate. Let the FM carrier be 10 MHz. Also, suppose the baseband signal is a pure tone at 1 MHz,

$$x(t) = \alpha \cos(2\pi f_m t)$$

where f_m is the frequency (1MHz) and α is the amplitude.

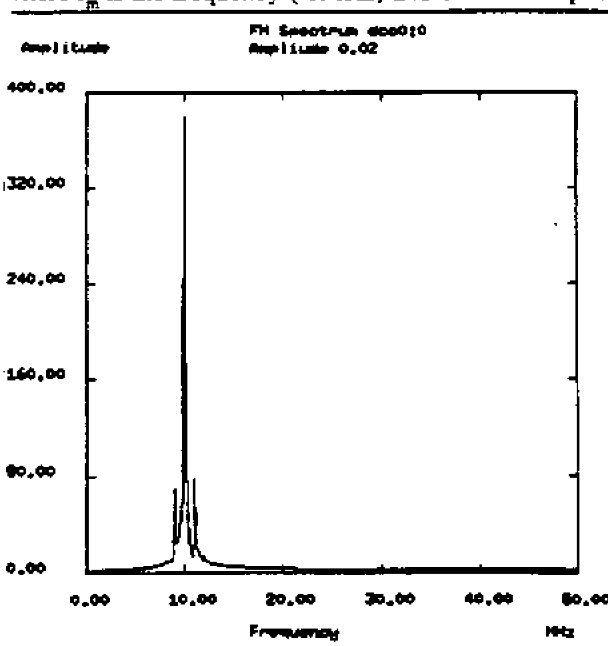

Figure 13. Spectrum, amplitude=0.02 (β= 0.32)

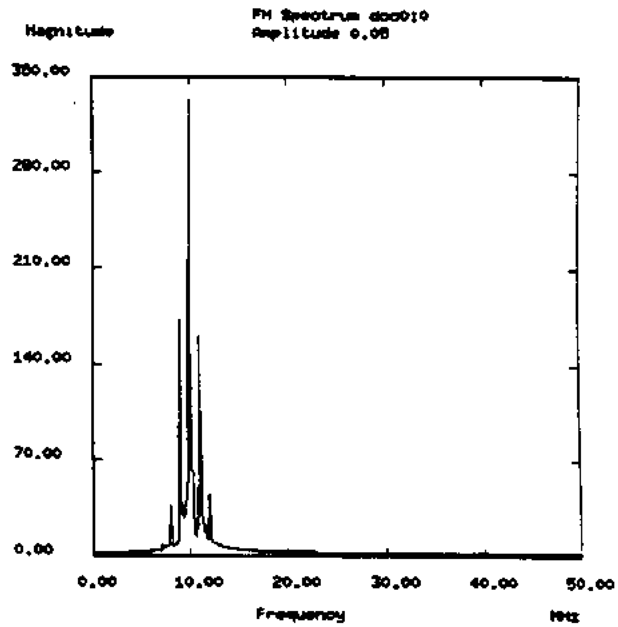

Figure 14. Spectrum, amplitude=0.05 (β= 0.8)

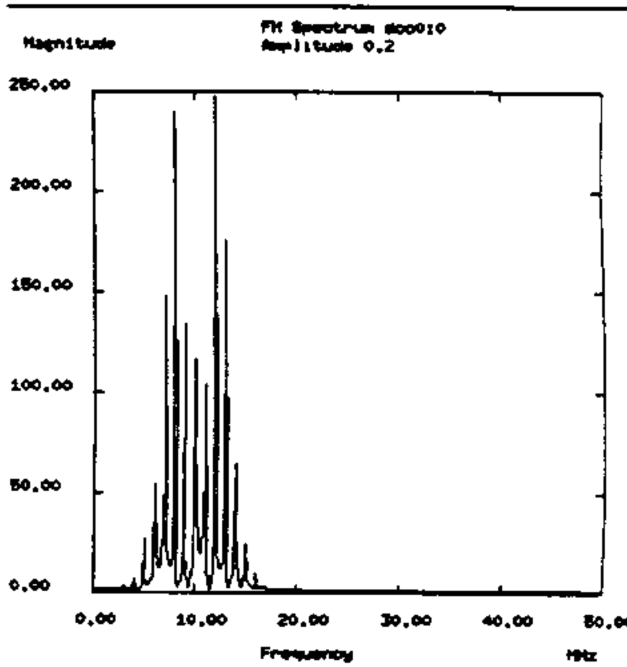

Figure 15. Spectrum, amplitude=0.05 (β= 3.2)

This tone is produced by the *sine* star in the topology shown in Fig. 12. Therefore, an appropriate sampling rate is 100 MHz. This rate can be decreased if the expected bandwidth of the FM signal is small. It must be increased if wideband FM is to be simulated.

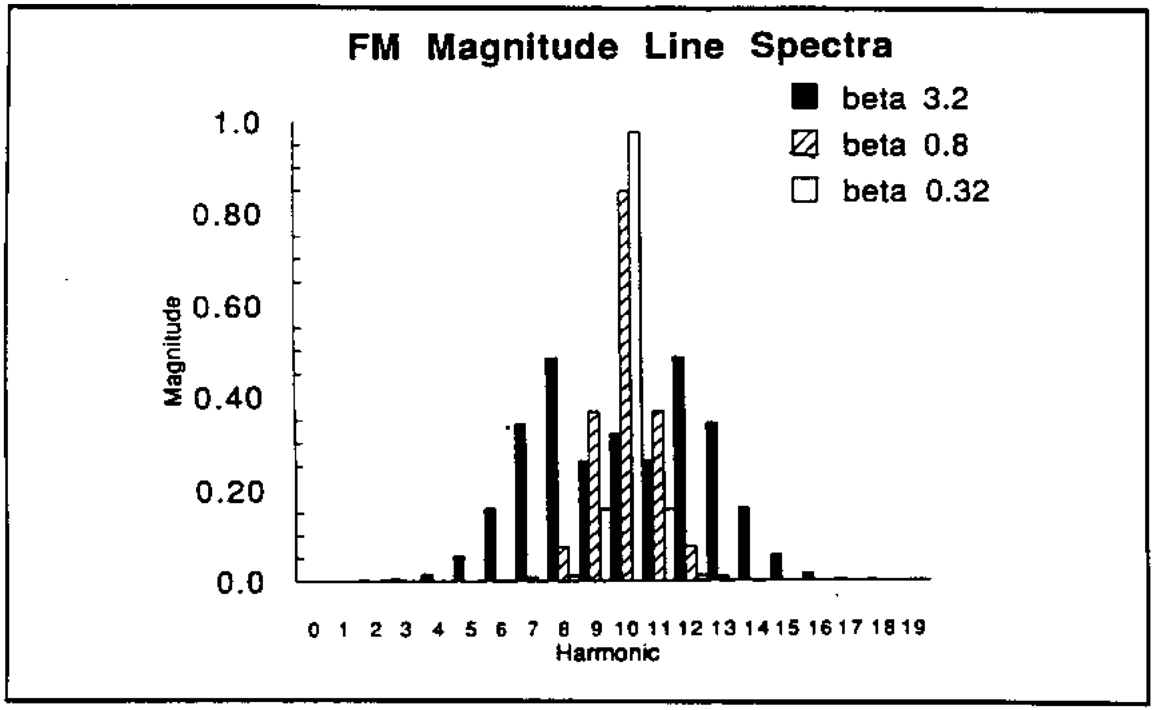

Figure 15. Theoretical FM Magnitude line spectra.

It can be shown that the FM signal in the time domain is,

$$\phi_{FM}(t) = A \sum_{n=-\infty} J_n(\beta) \cos(2\pi f_c + 2\pi n f_m)t$$

where,

$$\beta = \frac{\alpha}{2\pi} \frac{f_s}{f_m}$$

and

$$J_n(\beta)$$

is the Bessel function of the first kind.

Note how β depends on the ratio of the sampling rate and baseband frequency. Below, we will investigate the effects of increasing the amplitude a, and, therefore, β, on the spectrum of the FM signal

The spectrums of the FM modulated sine wave appear in Fig. 12, 13, and 14 for increasing amplitudes of the sine wave. Note that the magnitude line spectra of the FM signals follow the expected results from theory. See Fig. 15. As the amplitude is increased more spectral lines appear spaced at 1 MHz intervals. Also, the bandwidth increases. For low β, the signal is a narrowband FM waveform.

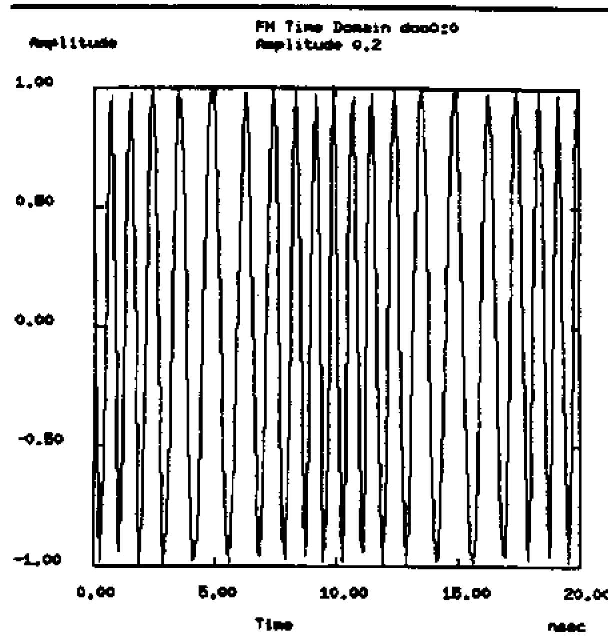

Figure 16. Time domain signal amplitude=0.2.

The FM signal for an amplitude of 0.2 is shown in Fig. 15. The demodulation of FM signals will be discussed in a separate application note dealing with phased locked loops.

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