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Topic	Student Name	Date	Course Leader
DSB-SC Demodulation	Wojciech Rościszewski-	01/05/2020	Dr. Inż. Łukasz
	Wojtanowski (ID:		Matuszewski
	140062)		

This laboratory exercise expands the characteristics of the DSB (double-sideband) linear modulation. Modulation is performed consistently to its requirements in order to achieve an accurate synchronization with the oscillating carrier. In fact, this can demodulated using MATLAB Hilbert transform script; however this isn't the practice of this experiment as here we will be using a program called Tina TI.

Introduction

Electrical signals can be found in almost all devices we all use ranging from telephones, TV's to radio broadcasts and so on. Therefore, within communication systems all of this information is being transmitted from one spot to another with the use of said electrical signals. Commonly message signals are not ideal to be transmitted, for example due to propagation qualities – note: only for a large wavelength. Usually most of these signals are at a similar frequency range therefore we must ensure that all of these transmissions are done at a different frequency in order to avoid any broadcast signal interference. We commonly use linear modulation, that translates the spectrum of message signal to a higher frequency, then this spectrum that is already translated is modified before it's transmission resulting in linear modulation schemes. Usually there are four, AM, DSB-LC, SSB, VSB and DSB. In this experiment we will simulate and examine the characteristics of DSB demodulation.

Below please find a block diagram of a balanced DSB modulator using AM modulator.

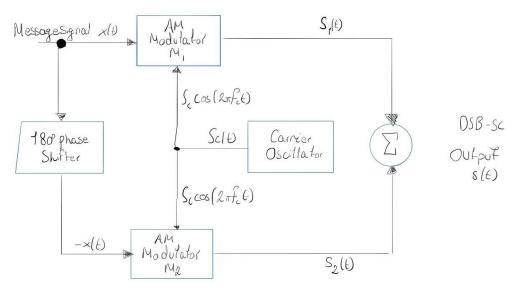


Figure 1 Presents a DSB-SC Modulator

We will be constructing something similar, only to pass a signal through it and be able to demodulate it.

Measurements

All measurements for this experiment were simulated using the program TINA TI where we will assemble circuits for the DSB-SC signal generation. All measurements for this experiment were simulated using the program TINA TI. Please find the equation describing the DSB signal wave below and is used to assemble the circuit to generate our DSB signal.

$$s(t) = A_m A_c \cos(2\pi f_m t) \cos(2\pi f_c t)$$

As seen in the figure below, I have added two controlled sources as requested in the instruction manual. According to the given instructions and parameters I have set the expression as CS1 = V(N1)*V(N2) and number of voltages to two entries, later I proceeded to insert the generator with given VG1 and VG2 parameters.

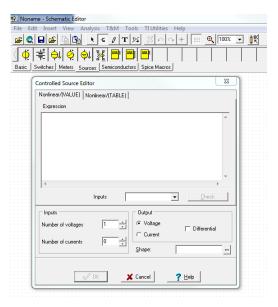


Figure 1.1 – Presents the process of specifying controlled sources.

Below please find the parameters that have been provided into the simulation for the voltage generators as given in instructions.

VG1 – Set to sine wave; amplitude of 5V; frequency 10 kHz (carrier)

VG2 – Set to sine wave; amplitude of 5V; frequency 1 kHz (message)

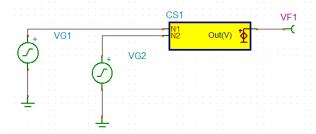


Figure 1.2 – Presents the generated voltage generator according to parameters provided. One controlled source and two voltage generators

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In the below please notice an exported version of the graph drawn by our graph with use of the Transient analysis mode set to 2ms as stated in instructions. As we can see we have an output signal generated that has amplitude of around 30V (see below figure 1.3 and find figures 1.4 and 1.5)

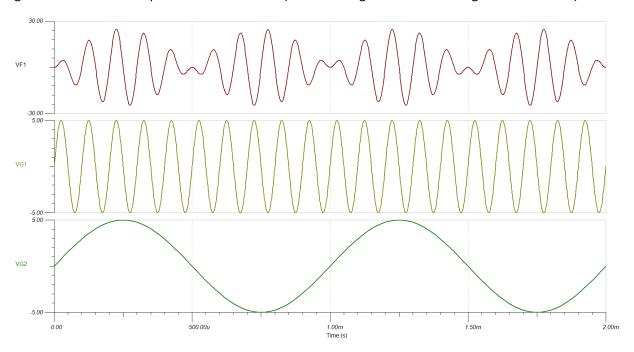


Figure 1.3 – Presents results of the transient state set according to provided parameters.

In the figures below please find the MAXIMUM and MINIMUM amplitudes of our output signal. To begin we will present the maximal amplitude value:

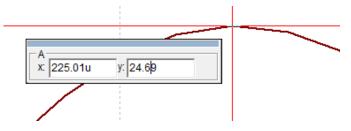


Figure 1.4 Presents the maximum amplitude of output signal

Below please find figure 1.5 presenting the minimal amplitude value.



Figure 1.5 Presents the maximum amplitude of output signal

Figures above only present and confirm that my circuit produces a viable sinusoidal wave as per provided parameters in manual, therefore we can continue the laboratory simulation.

presenting the circuit and design stages.

Moving on forward. We have successfully created a modulator, now we must design all components to design the demodulator, therefore we must firstly insert a new controlled source along with an additional voltage generator. As per provided parameters in the manual the expression of CS2 (controlled source 2) is set to V(N1)*V(N2) whilst the VG3 signal is set to a sinusoidal wave with an amplitude of 5V and a frequency of 10kHz (this is the heterodyne). Below please find figures

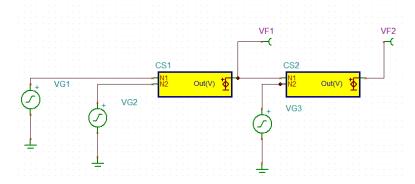


Figure 2 Presents new circuit with CS and VG added.

Moving on forward, we must not create a LPF (low-pass filter) using resistive and capacitive elements. Please find explanations of why these elements are used in the AM DEMODULATION laboratory report. Therefore, we add three resistors, and three capacitors in order to create a 2nd order LPF filter. Please find the figure below presenting such circuit.

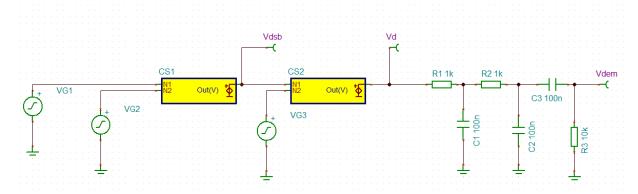


Figure 2.1 Presents new circuit with LPF.

Below please see figure 2.2, which shows the new output simulation (run at 4ms) where can see the changes that were made to our signal in comparison to figure 1.3.

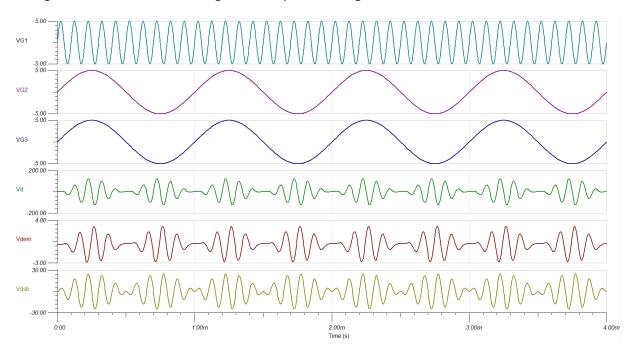
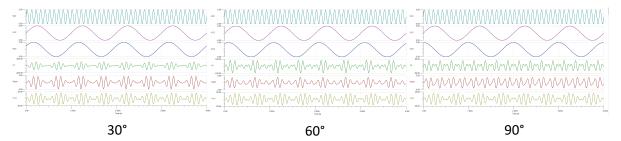


Figure 2.2 Presents simulation with circuit from figure 2.1

Moving on forward we will now change the VG3 by setting new parameters for the phase. Below please find 3 sets of figures. This will be done to ease presenting and comparing how changing the phase of VG3 can change our demodulated output signal.



Please notice the changing amplitude in the Vdem (V demodulated) signal. T The next part of our laboratory is to answer a question: Why there is the amplitude change when the phase changes? *Present appropriate equations. Please note that in our circuit we are working with several sinusoidal generators, which are multiplied. Since we are working with a periodically oscillating sinusoidal waves, we have to take into account the three fundaments of said signals, being: the amplitude, frequency and their phase. Please find the equation below that satisfies mathematical formulation of a sinusoidal wave $s(t) = A \sin(2\pi w_0 t - \varphi)$, where A is the amplitude, W_0 is the angular frequency and φ being the instantaneous phase. From this we know that number of cycles of a signal is dependent on frequency * time. For example, we have two sinusoidal waves in a frequency 200 Hz, they will get more and more out of phase between each other by 200 cycles per second as (cycles/sec)*sec is equivalent to the number of cycles in a signal. Therefore, a phase can only develop between two sine waves. Here we see that the sinusoidal waves are mutually shifted in phase. Here the mentioned waves are Vd and Vdsb as the two are multiplied by our controlled source. Whilst Vdem is the demodulated final output signal.

Below please find figures 2.6 - 2.8 these are exactly the same figures as figure 2.3-2.5 but at full scale for the sake of the accuracy of this laboratory report.

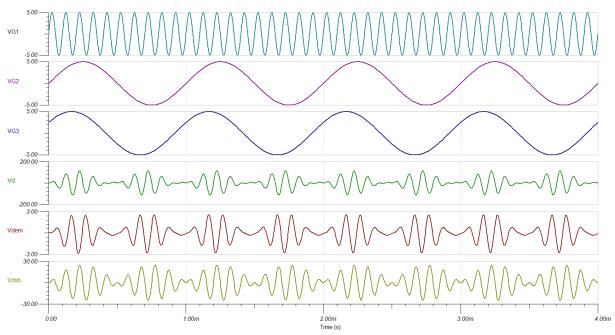


Figure 2.6 Presents simulation with VG3 heterodyne set to 5V, 10kHz with 30° phase shift

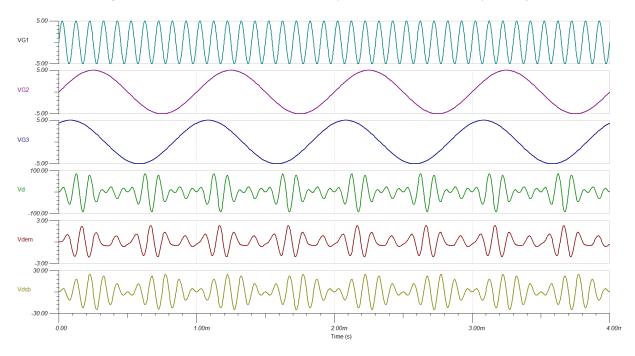


Figure 2.7 Presents simulation with VG3 heterodyne set to 5V, 10kHz with 60° phase shift

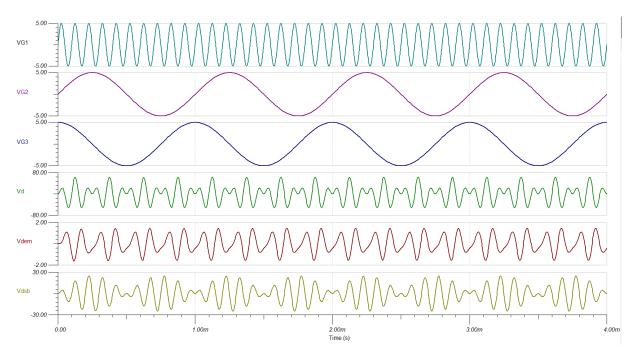


Figure 2.8 Presents simulation with VG3 heterodyne set to 5V, 10kHz with 90° phase shift

NOTE: Parameters of VG3 have been reset to 5V, 10kHz with 0° phase shift.

For the next part of our simulations we must now load VG2 with the included .WAV file prepared for this laboratory. Parameters of VG3 have been changed where amplitude is 1V whilst frequency is 10kHz.

Below please see figure 3.1, which shows the new output simulation (run at 2s) where the only changes made have been referenced in the above. The output signal shown is the Vdem curve being the demodulated .WAV signal.

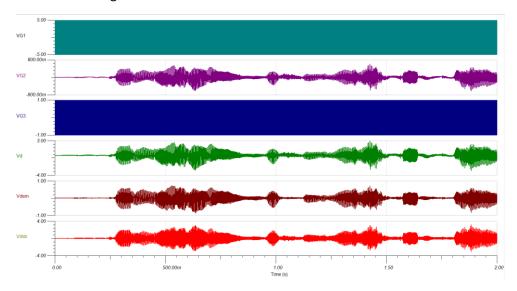


Figure 3.1 Presents simulation with VG2 set to .WAV file and VG3 set to 1V; 10kHz

From what can be heard is that the signal has a lot of interference, in a sense that it has changed into very high tones and can be barely understood. Either this is the fault that VG3 is causing this issue or the LPF is not filtering the correct high frequencies.

Now the VG3 parameters will be edited as before, where the phase has been shifted 30, 60 and 90 degrees. Run time of simulations has been set to 2s, below please find just similarly as in series of figure 2 the figures side by side as well as their full-scale copies.

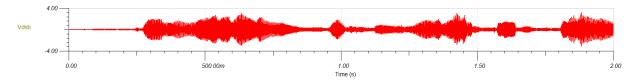


Figure 3.2 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 30° phase shift

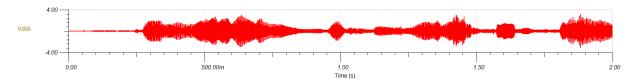


Figure 3.3 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 60° phase shift

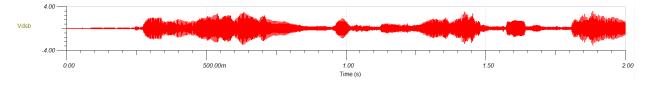


Figure 3.4 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 90° phase shift

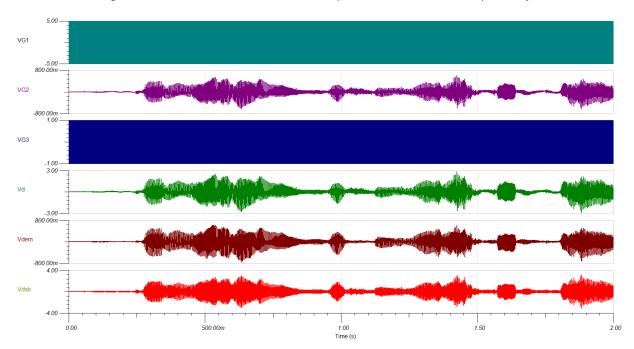


Figure 3.2 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 30° phase shift

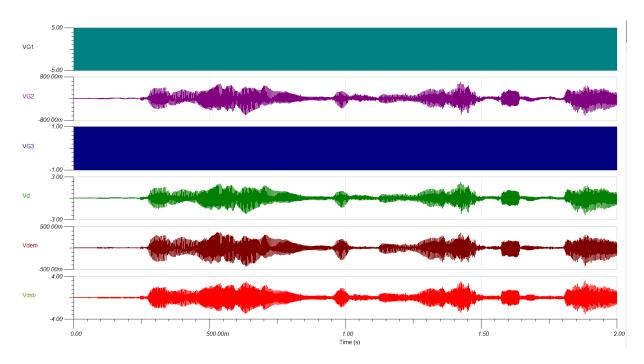


Figure 3.3 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 60° phase shift

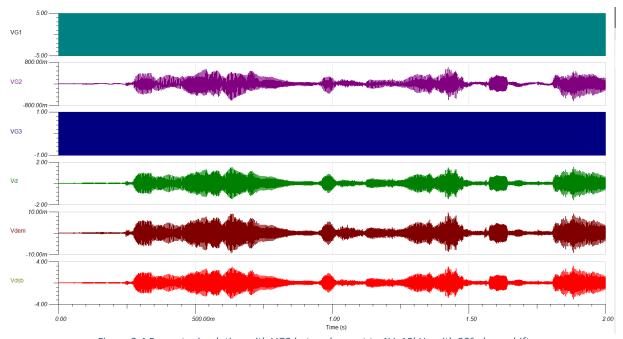


Figure 3.4 Presents simulation with VG3 heterodyne set to 1V, 10kHz with 90° phase shift

The higher the phase shift degree the more quitter the signal is. From this statement we can only infer that it synchronous detection is a powerful yet phase-sensitive technique of small signal recovery. I can only assume that this technique can be handy to recover signals with may have been obscured by some sort of interference which is far larger than the signal itself (so in this case our .wav is our signal of interest). To achieve a perfect result in using this technique we would have to create a band-pass filter that is very narrow in order to remove all sorts of frequencies excluding our frequencies of interest, however this can sometimes be practically impossible in an even when we would have to create an extremely narrow filter. In our example we are only filtering our the higher frequencies as everything below is of interesting mostly. After further research [1] I have confirmed

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from a technical article that we indeed are using a lock-in amplifier of sort, as we are using synchronous demodulation to move the signal that has been modulated whilst filtering anything else that is not synchronized to our referenced signal. From this we can only infer that as mentioned at the very start, when phase is shifted the signal is quite quiet however this is a false statement as the signal itself on the end (so coming from the speakers) is actually more and more quite but what is really happening is that the signal is not synchronized to our referenced signal (.WAV file in this case) therefore the signal is still the same but isn't in correct phase. To confirm this theory, I will change the both phases to be equal and see what will occur.

VG2 and VG3 have now been set to 1V amplitude and 10kHz frequency. Only factor that I am changing is the phase on one of the ends, firstly I am starting with 30-degree phase shift on both. Once done I will then change the phase shift on one of the ends to 90-degrees and see is signal will come out of sync and be quieter on the output. After simulation I have found that by changing the phase-shift of the signal we also change the amplitude of our demodulated signal to something much lower. Therefore, not only have I confirmed the fact that changing phase-shift affects the synchronization of the signals but also the amplitude of the signal being demodulated end is affected – the higher the phase shift in degrees the lower the amplitude. Concluding this, I can only assume that this is the 2nd order LPF causing this phenomenon however please only take this as an assumption as I am not fully satisfied that this is likely to be true yet no other explanation comes to me at the moment of conducting these simulations.

Moving on forward. Parameters of VG3 have been changed where amplitude is 5V whilst frequency is still 10.01 kHz and phase shift set to 0. VG2 has been changed to amplitude of 5V and frequency of 1kHz. Simulation has been run at 100ms.

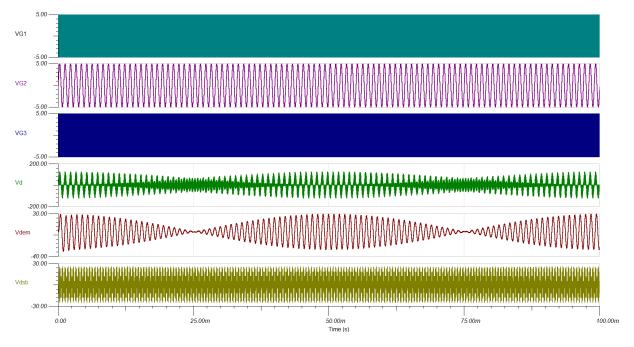


Figure 4.1 Presents simulation with VG3 heterodyne set to 5V, 10.01kHz with 0° phase shift

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Next step is to measure the period of Vdem envelope signal. From what we see the period is 75ms as seen in figure 4.2

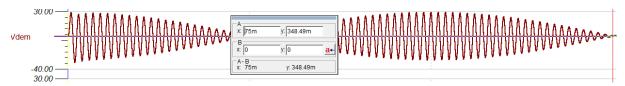


Figure 4.2 Presents simulation, with the period of Vdem.

Below please find simulation result from VG3 set to frequency of 1 kHz.

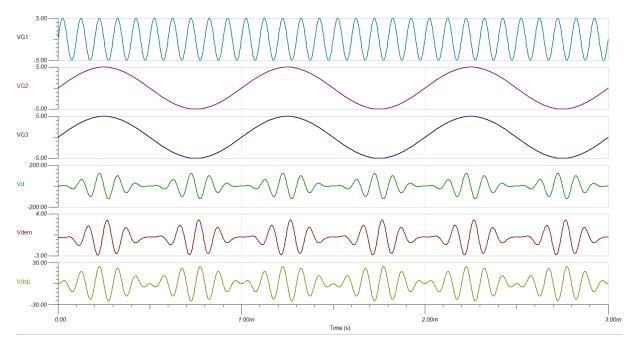


Figure 4.3 Presents simulation with VG3 heterodyne set to 5V, 1kHz with 0° phase shift

Period measured: 500.66µs

Below please find simulation result from VG3 set to frequency of 1kHz.

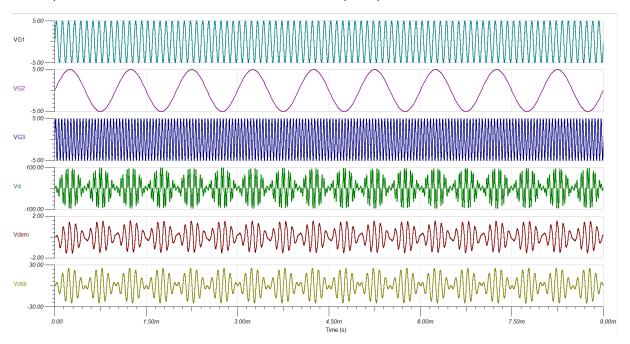


Figure 4.4 Presents simulation with VG3 heterodyne set to 5V, 20kHz with 0° phase shift

Period measured: 420µs

Below please find simulation result from VG3 set to frequency of 1kHz.

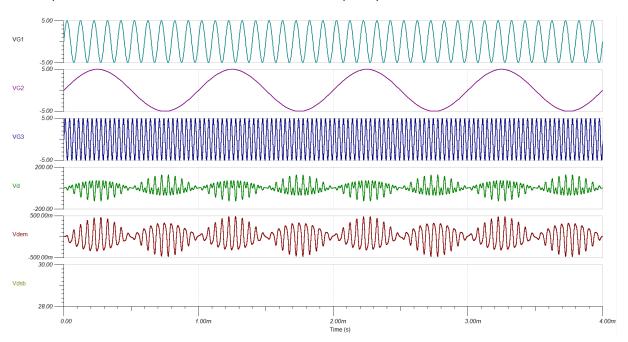


Figure 4.5 Presents simulation with VG3 heterodyne set to 5V, 30kHz with 0° phase shift

Period measured: 492µs

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From what I see the optimal frequency is 10.01kHz as that is when the period is quite large whilst going above or below in the figures above, we see that the period is very small. I can only assume that from this we could build an envelope detector. Considering our carrier frequency f_c and an envelope detector τ =RC. Therefore, the peaks of carrier can be achieved by formula T=1/ f_c .

The next steps of our simulations is to set VG2 to our included .wav file and VG3 set to a sinusoidal with an amplitude of 1V. Simulation repeated with VG3 frequency set to 10,01kHz. See figure below.

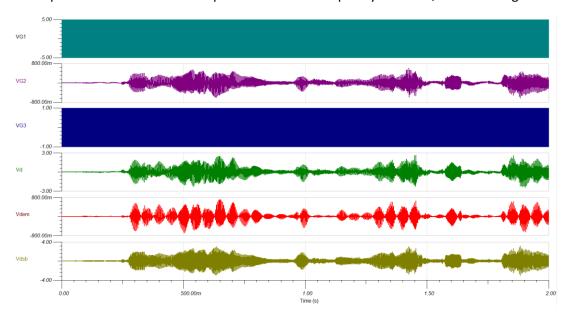


Figure 4.6 Presents simulation with VG3 heterodyne set to 1V, 10,01kHz and VG2 to .WAV file.

From the following simulations we see that our simulation follows a pattern. It's almost as if the input carrying the information is equal to the output of the signal. Therefore, the coherent detection is different from other demodulation techniques one important factor, that the output signals information are not deformed. However, we see slight signal deformation so I can only assume that the noise carrier always appears during demodulation in an additive manner – I say this because in some parts of the signals in figures 4.3-4.5 when zooming in we can see not only the fluctuations coming from our generator but also some additional signal which I think is just noise, and can only be seen to follow the fluctuation. Expanding this assumption further, it's like the additive noise follows our original signal hence why we have some deformations. Please treat this as only an assumption as I am not sure if this is correct and therefore cannot be treated as a true response.

Other demodulation techniques: In this experiment we have done synchronous detection demodulation on the receiving side where we have the received signal multiplied with a carrier frequency which has **the same frequency and phase characteristics** as our transmission carrier. This isn't a great option for demodulation, reasoning is because its very expensive as seen in the figures the circuits for this signal recovery process get extremely difficult and expensive. Asynchronous detection would be a better option, it is when our information signal is in the envelope of the receiving signal however if modulation of the signal is at 100% then demodulation with this method isn't the best option as it's not possible with asynchronous detection.

Bibliography

[1] - http://www.arrl.org/files/file/Technology/tis/info/pdf/9307028.pdf