Using ADALM-Pluto for DSB Monotone and Audio Signal Transmission/Reception

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***Abstract*—This project explores the modulation and demodulation of monotone signals like sin signals and sound signals using Adalm PLuto SOftware defined GNU Radio. It focuses on using the DSB technique to send and retrieve these signals, focusing on how to properly tune the components to ensure distortionless retrieval.**

I. Introduction

In communication systems, the transmission and reception of signals are fundamental to any efficient information exchange. Among a few, amplitude modulation (AM) is often used. DSB is the most implemented among several methods of AM since it uses bandwidth very efficiently by modulating both upper and lower sidebands of a carrier. The signal recovery then becomes difficult whenever there is much noise and other forms of distortions in nature. Software Defined Radios (SDRs), like the ADALM-Pluto, offer a flexible platform for implementation and testing of modulation and demodulation techniques. In this paper, a system using the ADALM-Pluto and GNU radio for transmission and reception of monotone and audio signals by DSB modulation is presented. The performance of the system will therefore be determined by the ability of the system to transmit and recover the signals with the least distortion, addressing issues of signal-to-noise ratio and optimization of modulation parameters. The results indicate that the system works satisfactorily under controlled conditions, but further improvements are needed for more robust communication in noisy environments.

II. Transmission and Reception

A. *Transmission*

Transmission is the fundamental process of propagating signals, or information from a source to a destination for different types of signals. The amplitude-modulated is the most straightforward way of accomplishing this by modulating the amplitude of a sinusoidal carrier wave using a baseband signal we call this signal Modulation.[4]

B. *Reception*

It is a simple method of recovering a signal from a modulated carrier wave that involves filtering out undesired interference and noise while keeping the modulating signal the signal could have coherent or non-coherent distortion while it is transmission the received signal remultiplied by the carrier and passed through a low-pass filter to recover the message signal. We call this Demodulation[2].

C. *Modulation and Demodulation*

AM(Amplitude modulation) it is a modulation technique that modifies the size of the carrier wave to transmit the signal, which is recovered after translated using lowpass , this type of modulation has types like normal and DSB modulation, the disadvantage of the Normal Amplitude modulation is that in the recovery of the message heavy noise will be found, also normal Am the maximum efficiency will be 33% since the carrier signal will take the most power for transmitted signal, however DSB modulation creates a signal with the main carrier frequency and two sidebands, which mirror each other the double-sidebands have the double bandwidth of the massage signal allowing us to reach 100% efficiency.[3][2]

III. Double Side Baseband Signal

The double-side signal it is a type of amplitude-modulated signal where it has mirrored upper and lower sidebands around the carrier frequency Only the two sidebands are transmitted which improves the efficiency of the transmission since the two sidebands are identical information, so it doubles the bandwidth requirement this type of signal is Complex to demodulate since it requires a coherent receiver to recreate the carrier signal any frequency or phase change leads to error in the demodulation.[1]

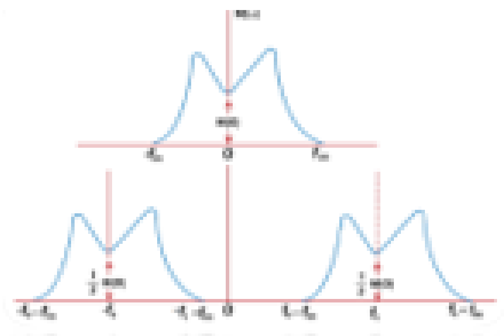

Fig 1. *Double Side BaseBand Signal*

IV. Systems Used For Modulation and Demodulation

A. *ADALM-Pluto (Software Defined Radio)*

It acts as a transmitter to send modulated signals such as AM. It can transmit custom signals using software like GNU Radio or MATLAB; these signals can then be demodulated and analyzed for various applications. [5]

Fig 2. *ADALM-Pluto Device*

B. *GNU Radio*

It is open-source software that allows the building of SDR (software-defined radios) and signal processing blocks. It is helpful in simulating the signal modulation and demodulation and testing it.[6]

V. System Design

The system was developed in such a way that it would Amplitude Modulate and Demodulate Double Sideband (DSB) signals with the help of GNU Radio. The system consists of two most important parts: Modulator and Demodulator. These components are very critical in the effective transfer and reconstruction of the wanted signal.[7]

A. *Modulation Process*

1. Reception of the Modulated Signal
The demodulation systems accept the modulated DSB signal from the modulated DSB signal passed through the ADALM- Pluto SDR.[11]

2.Multiplication with a Carrier Wave
The signal carrying the data is multiplied with the carrier wave which was generated locally and has the same frequency and phase as the original one. This reintroduces all the baseband components and the high-frequency components that were earlier suppressed.
3.Low-Pass Filtering
A low pass filter is placed in the multiplied signal to strip off components of high frequency (including harmonics, plus noise sources) so that signals at the low end, that is, at baseband are solely containing residues.
4. Signal Recovery[12]
The output that is filtered represents the recovered baseband output and in its form currently, it is consistent with the original modulating signal. It means if it was an audio signal it could be replayed or when it is data more computations could still be made.[13]

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VI. Experiment And Results

A. *Sine Signal*

First, The design of the modulator and demodulator for the sin signal was implemented. The output was examined however, it didn't match the input signal. To ensure accurate sampling a frequency of 100 kHz was selected because it's 4 to 5 times higher than the largest frequency which is the carrier, which guarantees that there is enough space to examine the signal without there being aliasing or oversampling.Additionally fc was chosen to be large enough to carry the signal but not too large to lose it. To match the Input The signal had to be run through a low pass filter with a gain chosen at 4. This gain was chosen to eliminate the values coming from multiplying the trigonometric functions which equals 1/4. The cutoff frequency which should be from fm to less than 2fc was chosen at 5k which is fm to eliminate all portions of the carrier signal and only keeps the input signal. The transition width was chosen at 250 because it determines how sharply the filter will cut off the frequency after the bandwidth. The output was observed to be the same as the input after these adjustments.

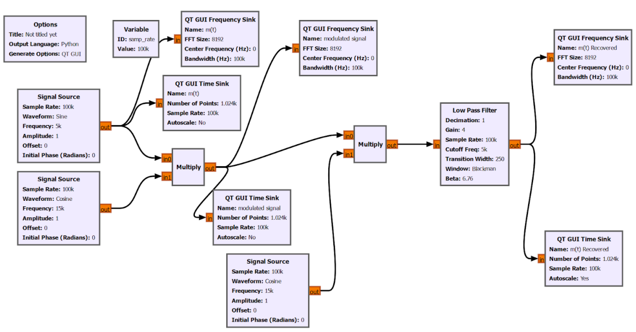

Fig 3. *modulation and demodulation of sin signal*

By examining the output we concluded that our design was correct since the input and the demodulated signal are the same.

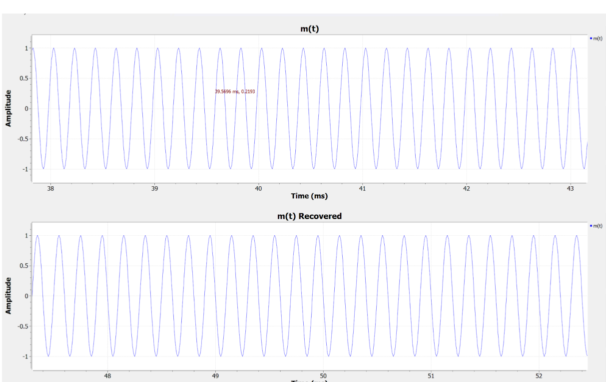

Fig 4. *Time Domain Waveforms of m(t) and the recovered signal*

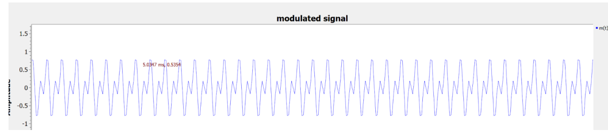

Fig 5. *Modulated Sine Signal*

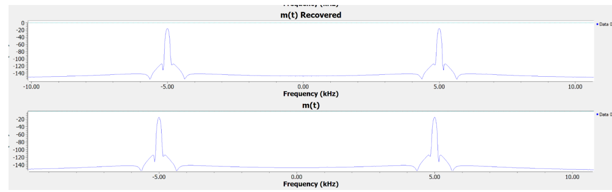

Fig 6. *Frequency Domain Waveforms of m(t) and the recovered signal*

B. *Audio Signal*

To modulate a sound source first it is needed to resample the audio source file. This is needed to ensure proper interpretation of the data. If the sample rate is much higher or lower than the file this will cause distortions in the signal.

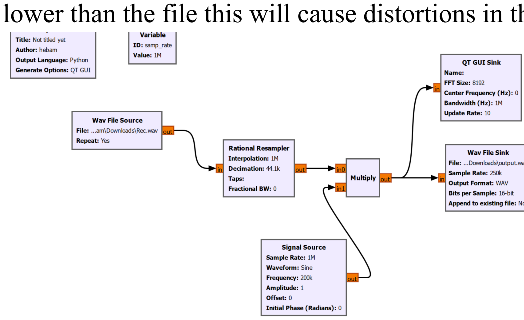

Fig 7. *Modulating Audio Source*

The Wav source file is sampled at 44.1K however the system's sample rate is 250K. The interpolation is used to upsample and decimation is for downsampling By multiplying the sampled file by 2.5k (interpolation) and dividing it by 441(decimation) the signal can now be properly carried in this system. The modulated signal is stored in a file to be retrieved by the demodulator.

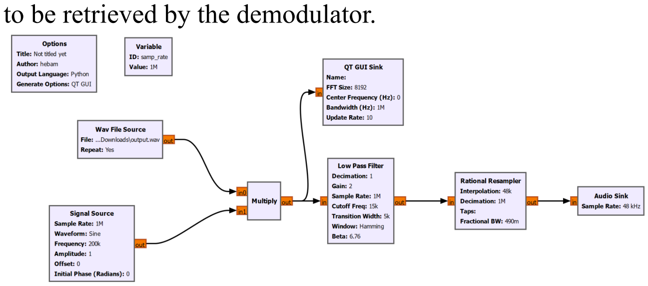

Fig 8. *DeModulating Audio Source*

The modulated signal is retrieved in the demodulating system and multiplied by c'(t) to demodulate it. Then to remove all unwanted signals and just keep the original signal it is run through the low pass filter. Since it is m(t)c(t)c'(t) the added value to the amplitude is ½. That is due to the fact that we only multiply two trigonometric functions, so the gain must be 2. To use the audio sink to hear the sound the sample rate for this block must be 48kHz. To achieve this the signal is rationally resampled 250k *24/125 = 48kHz (old sample rate = new sample rate * interpolation/decimation). Examining the output from the AudioSink the audio signal after demodulation is clear and without distortion.

C. *Audio with Pluto SDR*

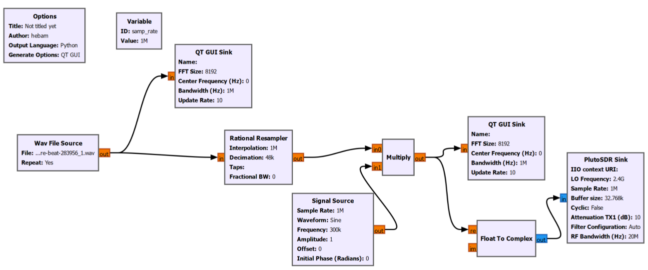

Fig 9. *Modulating Audio Source with Pluto*

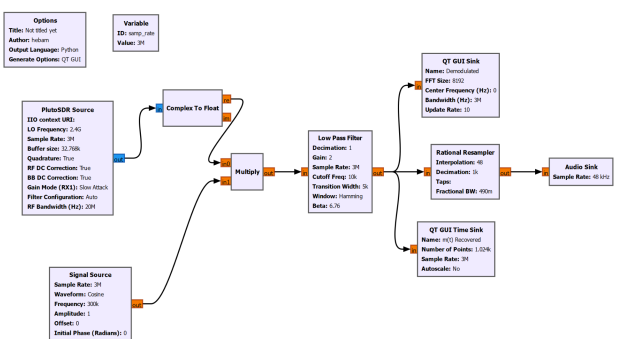

Fig 10. *DeModulating Audio Source with Pluto*

In our project, we sent the signal on ADALM-Pluto and received it on the receiver. During this process, we faced distortion on the receiver so we tried to decrease it by updating the values of the low-pass filter, sampling rate, carrier frequency, gain, and amplitude. After multiple attempts, we tried to evaluate the value for the cut-off frequency, transmission bandwidth, and sampling rate. So the adjustments are: sampling rate of 1 million, cutoff frequency of 10 kHz, transmission bandwidth of 5 kHz, and carrier frequency of 300 kHz.

After These adjustments the signal quality was improved and we had less distortion.

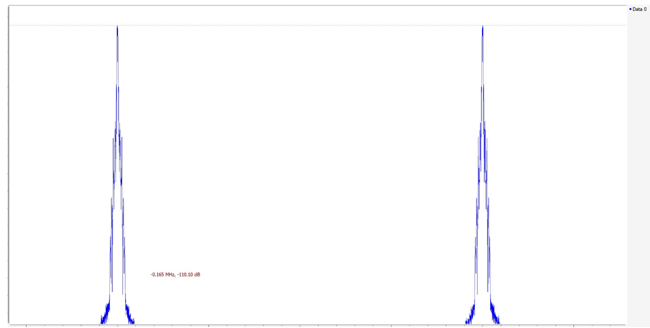

Fig 11. *Modulated Audio*

VII. Conclusion

Results obtained from the GNU Radio methodology show how signals are structured both in time and frequency. Examining the frequency analysis reveals some peaks, which mean the system has been doing pretty well in delivering the main features of a signal with hardly any distortion. On the time side, the output is periodic and unchanging in nature, just like the signal that was sent, meaning it captures the dynamics of the signal pretty well.

But yes, there it is a fair amount of improvement, yet performing decent in this case. We can add improvements in the form of better output controls to improve signal-to-noise ratios as dealing with noisy signals can impact output quality very badly. In addition, adjustments of the FFT parameters and appropriate windowing allow a proper view in the frequency domain to be achieved. We can then go further ahead by implementing higher-order modulation methods and also go for a range of operating frequency.

This will bring better performance and also provide flexibility in platforms for signal and communication systems.

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