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Design Analysis of Analog Data Reception Using GNU Radio Companion (GRC)

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Abstract: Optimization of the communication systems is one of the big challenges to the modern world. The problem is nicely dealt by the GNU Radio; a cheaper and powerful tool. GNU Radio companion (GRC) gives the simplicity to use as is the case with normal Python language used for programming. The Universal Software Radio Peripheral (USRP) gives the boost to GNU Radio capabilities. Software Defined Radio (SDR) is easily implemented by the combination of GNU Radio and USRP with its own ability of easier handling. Two techniques of Radio reception namely Amplitude Modulation (AM) and Single Side Band (Single Side Band) Modulation are compared and analyzed with the help of GRC in current work. The GNU Radio version 3.2.2 is used with USRP1.

Key words: GNU Radio · GRC · USRP · Amplitude Modulation · Single Side Band Modulation

INTRODUCTION

GNU Radio is software used extensively for the analysis and development of communication systems, more precisely known as Software Defined Radio (SDR). As compared to conventional radios, SDR gives us the benefit of diversity in radio specifications. GNU Radio comes handy in two ways. First is that it is a free ware software, made particularly for free operating system of UNIX, thus everyone enjoys its benefits. The second and more important one is that it is completely open source. So the experts from all over the world keep on updating it with latest ideas and extending beyond. Development is done in C++ which is a hardware level language; to achieve the best performance possible [1].

GRC (GNU Radio Companion) gives the pictorial view of the GNU Radio. Thus all the sources, sinks and processing blocks can be actually dragged and connected in a much easier fashion giving a complete grip on the problem [2]. Along with this fascinating feature, it gives proper graphical view of the signals in time and frequency domain making this software ideal for analysis. These capabilities are further enhanced by the use of USRP (Universal Software Radio Peripheral) device linking the real and digital worlds.

The USRP constitutes of FPGA (Field Programmable Gate Array) with daughter boards and Converters (DAC and ADC). Together in one assembly, they act as a transceiver and Up convert and Down convert the signal in transmitter and receiver respectively. These are generally called with different names as per their capabilities; USRP1, USRP2, e100 etc. Furthermore USRPs are classified on the basis of the operating frequency. Meeting the requirements of our current work, we use USRP1 with daughter boards LFTX for AM and basic TX/RX for SSB. The first one has a frequency range of DC to 50 MHz and the second one has 1 to 250 MHz [3].

Amplitude Modulation (AM) is a technique of transmitting information with the help of a radio carrier wave. It works on the principle of varying signal in accordance with the information that is being sent. In this technique, the carrier does not oscillate in amplitude by itself. However, the modulating data seems in the form of signal components, having the frequencies higher and lower than that of carrier. We can call these signal components as sidebands. The frequency bands which appear below and above than the carrier frequency are called lower sidebands (LSB) and upper sidebands (USB) respectively [4]. The fluctuations in the signal amplitude can be controlled by the sideband power.

Another technique used for radio transmission is Single Side Band (SSB) modulation, which is more bandwidth efficient. For a fixed bandwidth, twice the information is sent by SSB as compared to conventional AM. So the job is done by the transmission of one sideband only giving it the name SSB. We can generate SSB signal by the use of 3 methods; Sharp cut-off Filters, Phase Shifting networks and the Weaver's

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method. Most commonly used method is selective filtering method in which we use sharp cut-off Low Pass Filter (LPF) to remove undesired signal to generate SSB modulated wave. Ideal filters are required for these purposes which are unrealizable. However this is realizable if there is a separation between pass and stop band. Voice signal provides this separation as it has low power contents near origin and above 3.3 KHz. Its transition band is about 600 Hz.

System Description

Conventional AM Modulation: The AM transmission of different transmitting stations can be viewed by the use of a device named as Universal Serial Radio Peripheral (USRP). The date source of the USRP is one of the Sub Miniature version A (SMA) connectors, selected by the "Receive Antenna Setting". Basic LFRX uses the "RXAB", "RXA" and "RXB" settings. We can achieve 64 Mega-samples per second / decimation from the USRP source. The USRP is tuned at the frequency of 710 Khz for the purpose of receiving AM signals. The decimation rate of the USRP is set to 1 for the sake of simplicity with the 0dB gain [4].

We want to listen to the channel with the frequency of 790 KHz which is actually at 80 KHz in the FFT plot. Then applying the basic definition of the AM receiver, the USRP source is multiplied with the carrier of center frequency -80 KHz with amplitude of 1. The signal source in the block diagram is acting like a carrier having the default sample rate with amplitude of 1. The frequency of this signal source is selected by the variable slider block as shown in Figure 1. In variable slider we define a default frequency with upper and lower limit and it is actually controlling the frequency of the signal source.

The signal resulted after this multiplicative action is passed through a low pass filter to reject the unwanted channels. The filter used in this case is a FIR filter. The parameters of the low pass filter are set to fetch the required information. The AM broadcast signal has the bandwidth of 10 KHz (+/- 5 KHz from the carrier frequency). Hence we choose the cutoff frequency as 5 KHz with the transition width of 100 Hz. The sample rate is decimated to 64 KHz for better resolution of the received signal. The variable block in Figure 1 is the controlling factor for the decimation of the sampling rate [5].

The windowing function [6] of filter can be described as the mathematical function which gives the output in some interval and zero outside. The purpose is to reduce the leakage of the spectrum but somehow it manages to change the shape of the leakage. There are many windows which affect the shape of the spectrum and we choose according to the application. For example some provides service for frequency resolution, some make it comfortable to detect the desired frequency and some helps in the amplitude accuracy. The window chosen in the current work is Hamming Window and the beta factor shown in the block is only applicable for the Kaiser Window.

After the low pass filter, we have the block of Automatic Gain Control (AGC). The main function of this block is the feedback control mechanism for the adjustment of the gain to a certain level of interest. In the case of AM receiver the sound is varying rapidly between low and high levels, it adjusts the volume accordingly.

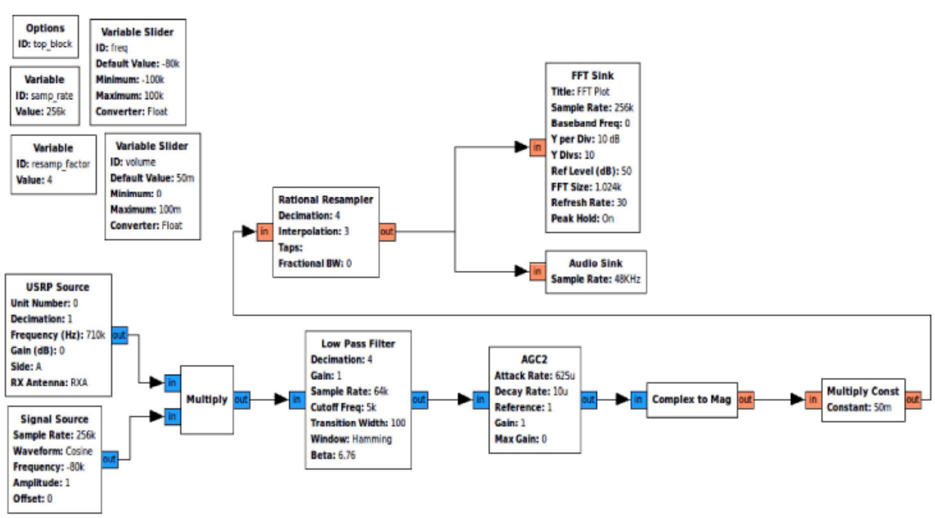

Fig. 1: AM Receiver

The attack rate of the AGC can be defined as the how quickly AGC decreases the gain when we receive a loud signal. For this case it is chosen to be 0.000625 so that it can adjust the gain to an appropriate level. In the same way decay rate of the AGC can be defined as the how quickly AGC increases the gain when we receive the weak signal. For this case it is chosen to be 0.00001 so that it can control the gain to a certain level. Reference level of the AGC can be defined as the level that AGC tries to maintain. In our case its value is 1. Maximum gain of the AGC is the gain that an AGC can have, which is equal to 1 in current case.

For AM the baseband signal is in the form of magnitude of the modulated waveform. To achieve this type of signal we use the “complex to mag” block, which accepts the input as complex and gives a single float value. Then we have a “multiply constant” block which is for the volume adjustments. Slider display of the volume is actually for this purpose. In another variable slider block, shown in Figure 1; the default value is set to .05 with the upper and lower limits to .1 and 0 respectively [7].

We have decimated the sample rate to 64 KHz from 256 KHz for the purpose of viewing better signal. If we want to hear the radio channel then we have to adjust this sample rate to a value compatible with the sample rate of sound card of the system. As the sample rate of the audio sink is 48 KHz, so we have to convert our 64 KHz to 48 KHz [8].

This can be done by the use of “Rational Re-Sampler” block. The decimation value is set to 4 and hence we choose interpolation as 3 to achieve the required sample rate. The other parameters of Rational Re-Sampler are left empty for automatic adjustments.

At the end the output can be listened through the “Audio Sink” block. In this block we only define the sample rate as per acceptable by the sample rate of sound card of the system. In addition to the audio sink we can have the block of FFT sink for the graphical view of the desired channel. In the FFT block we define the reference level as 50 dB and FFT size as 1024.

Analysis of Conventional AM Reception: The FFT plot of the received signal shows multiple peaks with varying amplitudes. These peaks are the indication of placement of carriers that corresponds to AM broadcasts and the varying amplitudes are because of live transmission. The sidebands for the stronger waveforms can also be observed in Figure 2. The frequency slider can be varied to hear other stations. Our required signal of 80 KHz has been shifted to origin as displayed from the view graph. Figure 3 depicts the output after the low pass filter. The required channel is tuned and the other remaining channels are discarded. Now this signal will be processed further to be converted into magnitude form. The magnitude form of the modulated waveform for the baseband signal is shown in Figure 4. The bandwidth of 5 KHz can also be observed in Figure 4. This signal is fed to the rational re-sampler for decimation according to the audio sink's requirement. This final signal which can be heard through the speakers is shown in Figure 5. The signal is now decimated to 48 K Hz, as shown in Figure 5.

Single Side Band (SSB) Modulation: USRP source is tuned at 50.3 MHz giving complex output with decimation of 1 and 0 dB gain. This signal is fed to the low pass Xlating FIR filter (Translating Filter) set at center frequency of -51.5 KHz. Xlating Low Pass Filter is used to shift the whole spectrum to origin and then discard the undesired side bands to achieve our required station. The center frequency will shift down the spectrum by 53 KHz and the cut off frequency is 2 kHz with transition width of 100 Hz having gain of 0 dB.

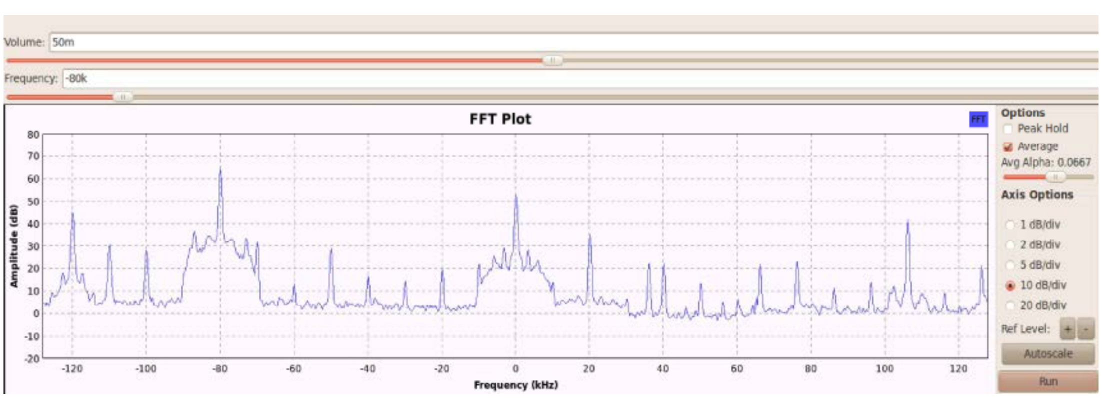

Fig. 2: AM Broadcast Reception

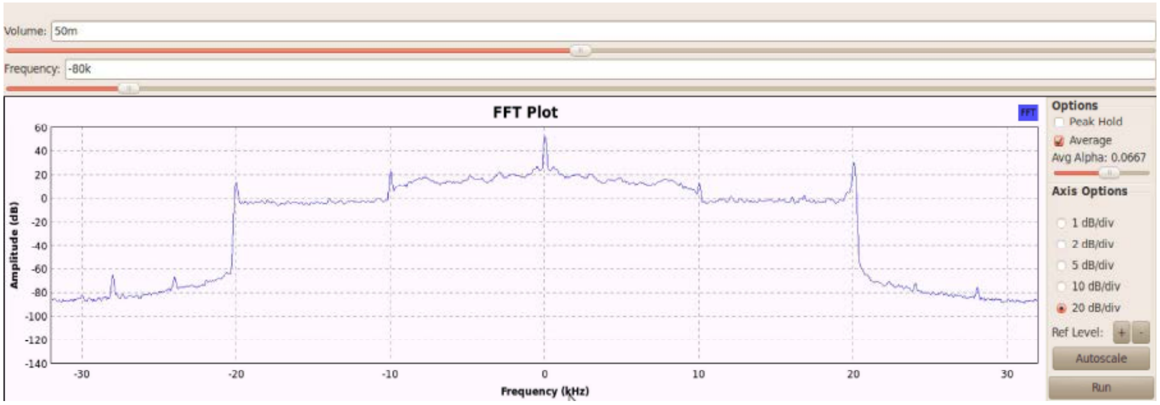

Fig. 3: Low Pass Filter (LPF) Output

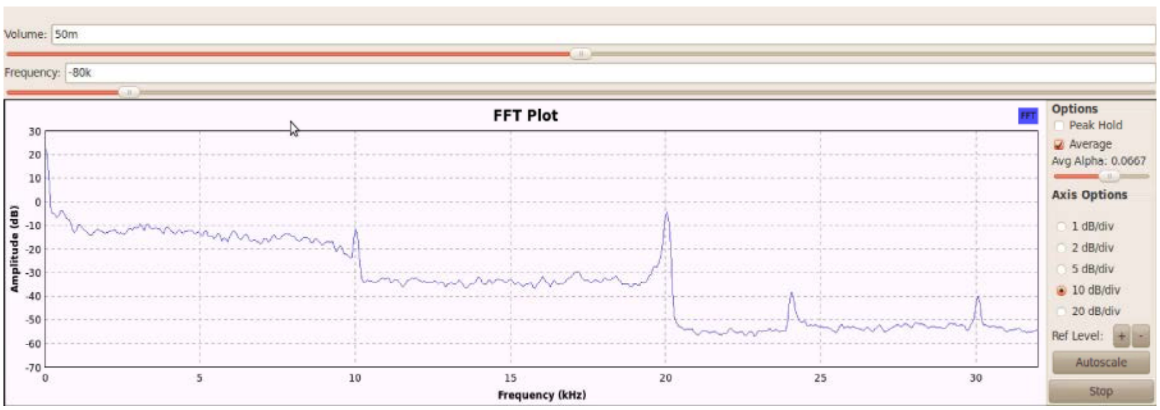

Fig. 4: Baseband Signal in Magnitude Form

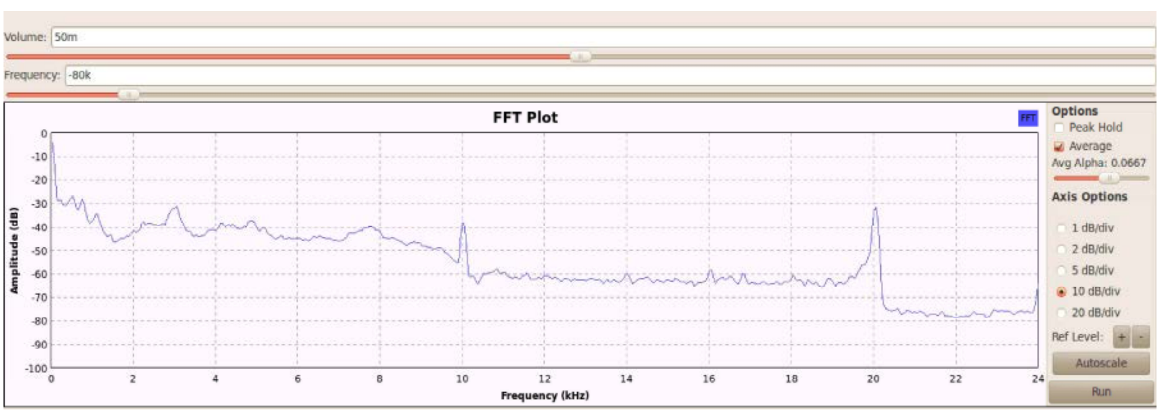

Fig. 5: Final Output

For Weaver method, the complex signal generated by the USRP is converted to real and imaginary parts by the use of "complex to float" block. This action makes it different from other SSB techniques. Real and Imaginary part of this signal is multiplied with cosine and sine signal source respectively having frequency 1.5 kHz. After multiplication we get our spectrum shifted right and left by 1.5 kHz whose output type is float and then both the parts are added. Before rational re-sampler we use throttle, this is used specifically in GUI applications with no rate limiting block to keep processor out of congestion. Throttle block doesn't control sampling rate, which is controlled by the source or sink, but it only stream lines the data which is easier to handle by the processor.

Rational Re-sampler is used for down-sampling (decimation) and up-sampling (interpolation). We use decimation factor of 16 and interpolation factor of 3 mean we down sample our signal with a factor of 16/3 (now sample rate is 48 KHz from 256 KHz) to keep sampling frequency within the range of our audio device as well. This is a necessary step as the sound card doesn't work at every sample rate. We leave the taps and fractional bandwidth empty. The rational re-sampler can be used as a filter if intended, but leaving the taps empty means that it is used as an all pass filter. Thus no effect on the signal is observed. We are listening to this signal on the speaker which is controlled by the sound card used in the system having range from -1 to 1 but we are getting signal of very high amplitude so we do multiply it with a variable source with a default value of 50 µ. It can be adjusted to a value where proper audio is observed. Number of inputs is set to two to get a good sound quality from stereo speakers.

Analysis of SSB Reception: We execute the flow graph and took its FFT plot just after USRP source, 256 kHz wide spectrum was seen as this was sample rate. There is a signal between 40 and 60 kHz as shown in Figure 7. USRP was set at 50.3 MHz frequency so the signal at 0 Hz was seen at 50.3 MHz. By moving the cursor of its FFT plot, it was viewed that its right edge frequency is about 53 kHz, this is the carrier frequency because 0 Hz signal is at 50.3 MHz so the original frequency will be 50.3 MHz + 53 kHz = 50.353 MHz.

To get the required signal, whole spectrum will be shifted down by 50.3 MHz and then extract it after passing low pass filter. For this Xlating low pass FIR filter is used having a center frequency of -51.5 KHz. This gives complex output having spectrum width from -1.5 kHz to 1.5 kHz as shown in Figure 8. To remove left side band of the spectrum, we break the signal into real and imaginary parts by using complex to float block and then we multiply its real part with a cosine signal and its imaginary part

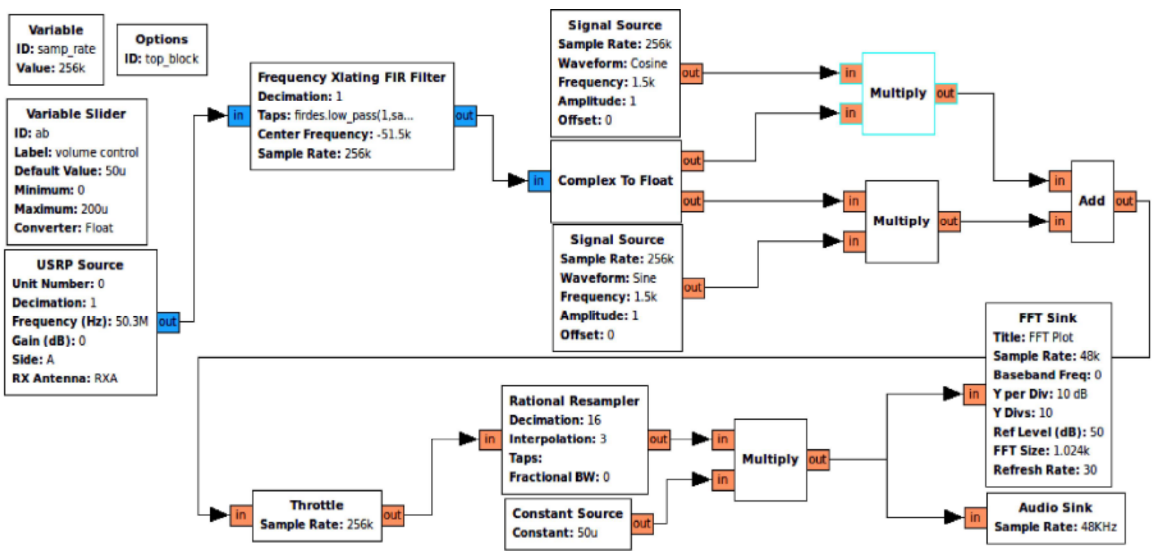

Fig. 6: SSB Receiver

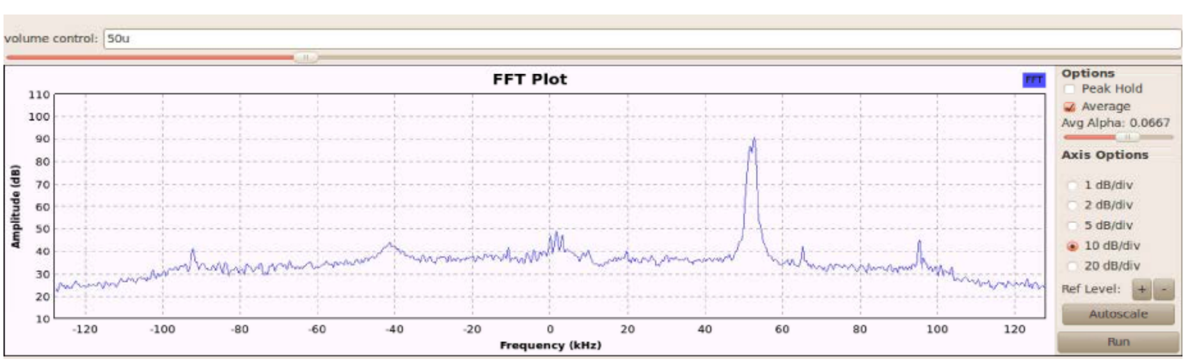

Fig. 7: Input to the System

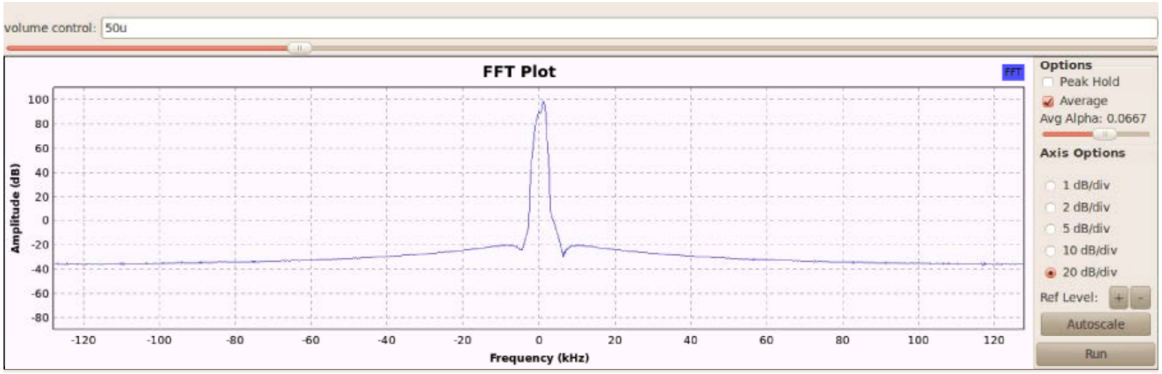

Fig. 8: Output of the Xlating Filter

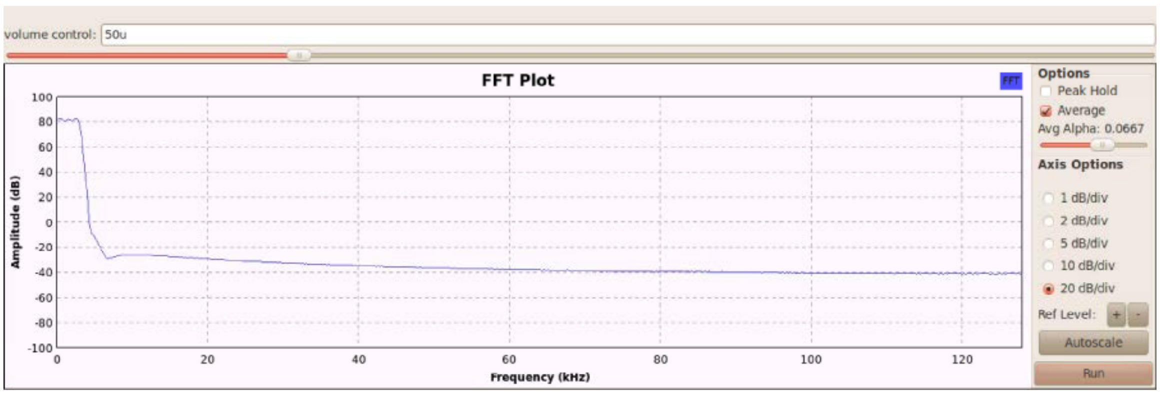

Fig. 9: Input Signal to the Adder

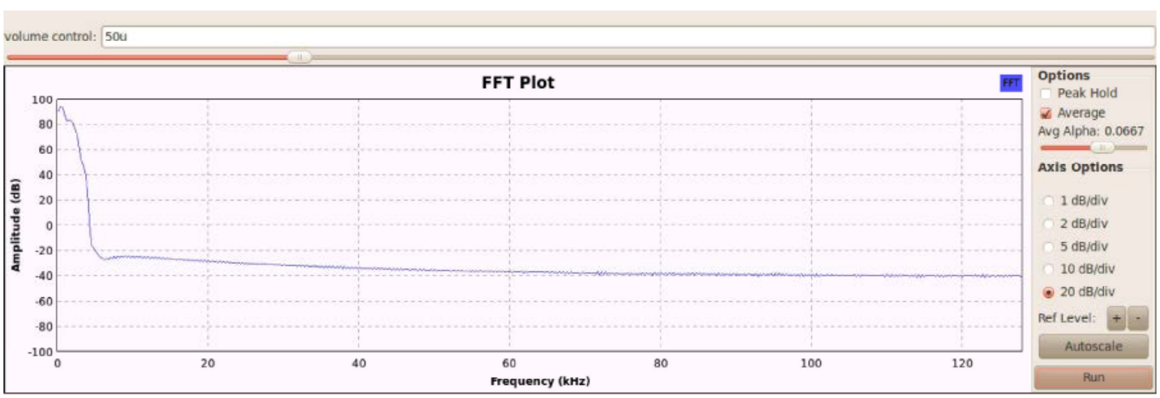

Fig. 10: Output of the Adder

with a sine signal of 1.5 kHz frequency. Amplitude of both signals which we receive after multiplication is about 80 dB as shown in Figure 9. After addition of both signals, its amplitude is increased by 3 dB, so we get signal of amplitude 83 dB as shown in Figure 10. The speakers of personal computers have sample rate of 48 kHz and amplitude of -1 to 1 while the signal which we are getting is sampled at 256 kHz having amplitude from -50 dB to 84 dB. To get required sample rate of 48 kHz, used rational re-sampler with its decimation of 16 and interpolation of 3 and to achieve the required amplitude, multiplied with a constant source of 50 μ giving a signal of amplitude about from -140 dB to 0 dB acceptable by the speaker.

CONLUSION

AM and SSB are two different techniques achieving the same goal. It is seen through the diagrams implemented above that SSB is much more complex than the conventional AM technique. Though at the cost of this complexity, it gains some added advantages. The foremost is bandwidth which is about half of the AM. This bandwidth reduction has a direct effect on the coverage of SSB. SSB covers smaller band in the spectrum, so the same energy distributed over the small band gives higher power. Thus coverage is increased. GNU Radio opens the new door to the signal processing and communication advancements.

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