ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 4, Issue 6, June 2017

Real Time Voice Transmission using GNU Radio

[1] Pradeep H S
Dept.of ECE, Siddaganga Institute of Technology, Tumakuru, Karnataka

***Abstract*: Today the world is using the most advanced technologies everywhere. From electronic devices used for household to artificial intelligence there is a need for communication. Two devices may indulge in a communication where one has to transfer data while the other needs to receive for the operation. The problem becomes evident if the systems are at a very larger distances. So we need to go for wireless communication. Implementation of a transmitter or a receiver using the hardware becomes very complex and is indeed expensive. Debugging the hardware becomes a tedious process once the design gets implemented. Therefore, a need in reducing the complexity of hardware arises. This requirement enables software radio function with minimum hardware dependency. This paper aims at transmitting voice on a wireless media via Software Defined Radio (SDR) called GNU and Universal Software Radio Peripherals (USRP).**

***Keywords*— Software Defined Radio (SDR), GNU, Universal Software Radio Peripherals (USRP).**

I. INTRODUCTION

All electronic gadgets, say mobile phones, computer, projectors and many more need data to be received to operate. For example computer receives data from modem to connect to a network, similarly mobile receives data from the antenna to connect to another mobile. These are all possible only with the advent of wireless communication. Thus wireless communication is achieved using SDR. One such SDR is GNU radio.

Software Defined Radio (SDR): SDR is a radio communication system where components that have been typically implemented in hardware are instead implemented by means of software on a personal computer or an embedded system. SDR are used for its simplicity and exibility. With the right software, a single SDR chip could perform the functions such as recording FM radio and digital television signals, read RFID chips, track ship locations, or do radio astronomy which makes SDR versatile.

What is GNU radio? GNU Radio is a free and open-source software development toolkit that provides signal processing blocks to implement software radios. It can be used with readily available low-cost external RF hardware to create software defined radios, or without hardware in a simulation like environment. It is widely used in hobbyist, academic and commercial environments to support both wireless communications' research and real world radio systems.

II. LITERATURE REVIEW

Many techniques were developed in recent years. The main approaches of the techniques was to transmit and receive data without much hardware needed using GNU radio. In this section, a brief literature review is given.

In paper [1], the authors discuss about the emerging trends of SDR for the applications in modern communication systems.
In paper [2], the authors give the challenges faced while transmitting voice over wireless medium and discuss the solutions to that. This paper depicts how to transmit real time voice over wireless medium.
In paper [3], the authors focus on SDR from a discrete time sampling perspective and discusses the e orts that are currently being pursued in order to further bridge the gap between these discrete time samples. Highlighting the advantages and current issues with SDR technology, this paper also presents several examples using a recently released, commercially available SDR platform.
In paper [4], the authors portray the exibility and reusability of SDR. SDR designers have turned to FPGAs to provide a exible and reconfigurable hardware that can support computationally intensive and complex algorithms which can be used in a multitude of voice, data, and multimedia applications.
In paper [5], the authors discuss the flow graphs built using the blocks such as USRP sink and source, UDP sink and source, TCP Sink and source which corresponds to communication using antennas, D-link switches or modems.

III. SYSTEM OVERVIEW

Software Defined Radio(SDR): As the name suggests, a software defined radio is a reconfigurable radio whose physical layer functions are mainly or fully defined by software instead of hardware. Because of their exibility and cost eficiency, SDRs allow the implementation of radio communication systems that are interoperable among different standards, protocols, frequency bands, user requirements and functionalities. The Fig.1 is the overview of the system set up for the transmission and reception of data.

IFERP
ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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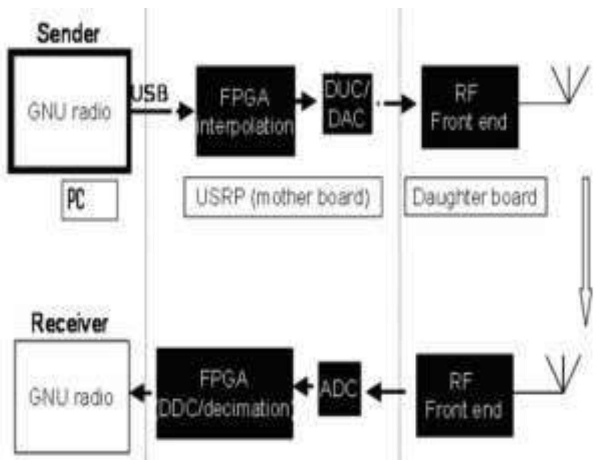

Fig.1: Overview of the system.

It is expected that, due to their adaptability, cost and time efficiency, SDRs will find widespread applications in mainstream telecommunications markets and become the dominant radio communications technology in the near future. A typical block diagram of SDR is as shown in Fig.2. The Universal Software Radio Peripheral (USRP) is a hardware platform that can be programmed into a software defined radio. The motherboard of an USRP provides all the functions and modules needed for signal processing. The platform can be programmed by an open-source software application called GNU Radio Companion (GRC)

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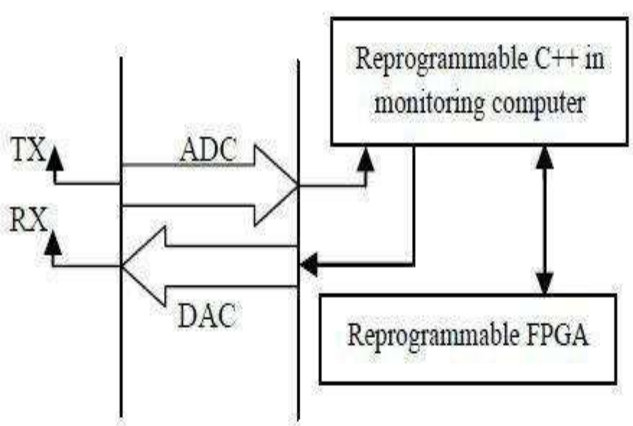

Fig.2: Block diagram of a Software Defined Radio.

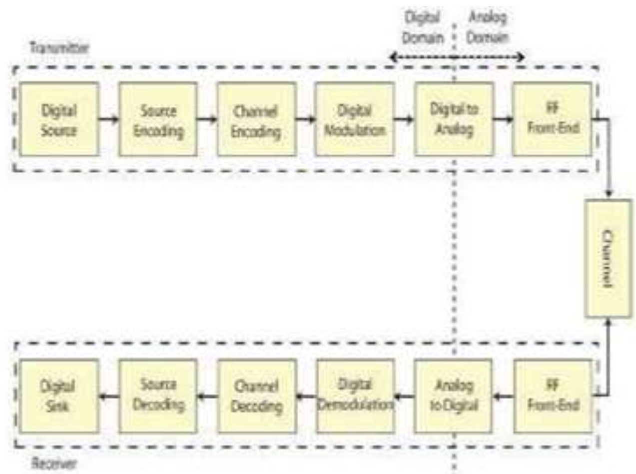

Fig.3: Block diagram showing the digital and analog divide in a Software-Defined Radio Platform

further into the RF block, programmability could be extended to the RF front end and an ideal software radio could be implemented. The advantage of having components implemented in software is exibility, as different frequency bands, air interface protocols, and functionalities could be upgraded through a software download instead of having to completely replace the hardware.

Thus, the ultimate goal for software defined radio is to move the AD/DA conversion as close as possible to the antenna so that all signal processing can be done digitally. However, some technical limitations make it currently infeasible to perform the AD/DA conversion at the antenna. Fig.3 shows the functional blocks that can be implemented in the digital domain of a communication system. These blocks include modulation and demodulation blocks, which perform mapping between bits and electromagnetic waveform characteristics; coding/decoding blocks, which help mitigate impairments in the wireless channel; source encoding and decoding blocks, which remove redundant information from the binary data; and channel encoding and decoding blocks, which introduce redundant information to protect transmissions from potential errors.

a. Digital Domain:

In an SDR platform, all of these components are implemented in software and can be run in different processing venues

ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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including Field Programmable Gate Arrays (FPGAs), Graphics Processing Units (GPUs), Digital Signal Processors (DSP), General Purpose Processors (GPP), or a combination thereof. While FPGAs are computationally powerful, they are relatively power efficient but in exible, and it is difficult to implement new modules in them. Similarly, GPUs are very computationally powerful but are difficult to use, especially when trying to implement new modules.

DSPs are processors that perform specialized mathematical computations. While users can implement new modules into them with relative ease, and they are relatively power efficient, they are not well suited for computationally intensive processes and can quickly lose speed. Finally, GPPs are a popular solution for SDR implementations and prototypes due to their high level of exibility with respect to reconfigurability. However, since GPPs are not specialized for mathematical computations, they can be very power inefficient.

b.Analog Domain:

The Fig.4 shows a typical RF front-end responsible for processing the analog portion of the digital transmission. In the transmission signal path, the digital samples are converted into analog signals by the DAC to be input to the RF front-end. The analog signal is later mixed with high frequency carriers and modulated to a determined RF frequency and transmitted over the air.

In the receiving signal path, the RF signal is captured by the antenna and brought back to baseband to be processed by ADC. The RF mixing and modulation is driven by the local oscillator (LO), which generates the RF signal, which is mixed with the incoming signal. Another very important component used in radio transmission is the Low Noise Amplifier (LNA), which is usually located close to the antenna and is used to amplify weak signals without significantly increasing noise level.

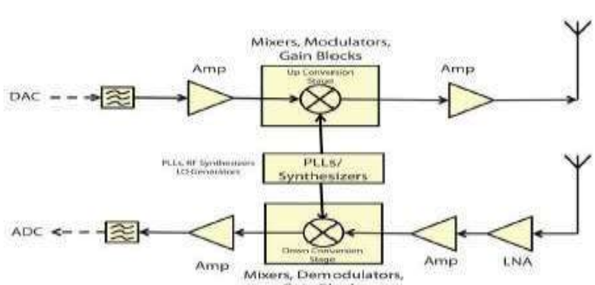

Fig.4: Block diagram of a typical RF front-end.

C. Sampling and A/D-D/A Conversion:

Digital transmissions are all about sampling. A continuous-time signal can be converted to a discrete-time signal using sampling, and a discrete-time signal can also be converted to a continuous-time signal by reconstruction. To sample a signal, instantaneous measurements are taken every Ts seconds. In this case, Ts is the sampling period, and fs = 1/Ts is the sampling frequency.

To reconstruct the original signal from the sampled signal, it is necessary to apply a low-pass filter on the sampled signal. The components responsible for sampling and reconstructing the signal are the DAC and the ADC.

As previously mentioned, moving the A/D and D/A conversions closer to the antennas is the ultimate goal of software defined radio technology. In order to do so, the major challenge lies in the DAC and ADCs sampling capabilities. To digitize an RF signal it is necessary to sample it atleast at the Nyquist frequency, and the higher the data rate of the signal, the higher the resolution required to capture the information. For example, an 802.11n Wi-Fi channel is 40MHz wide, which means the ADC has to digitize 80MHz of signal bandwidth, resulting in a sample rate of atleast 160 million samples per second (Msps).

d. GNU Radio:

GNU Radio is a free software toolkit licensed under the GPL for implementing software defined radios. It provides means for performing the digital signal processing portion of a communication system design. Several software algorithms include filters, channel codes, synchronization elements, equalizers, demodulators, decoders, and many other elements. It is possible to use these components as building blocks of a communication system, GNU radio not only provides these blocks but also a method of connecting them together. Communication systems can be implemented by using the already available blocks or by developing new components for the software platform. Basically GNU radio is associated with two languages, C++ for creating signal processing block, python for connections and generating a signal flow graph as seen in Fig.5.

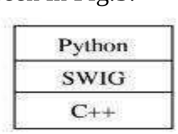

Fig.5: Layers of a GNU Radio Block.

ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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A signal processing block is typically a C++ function which will perform any processing on the input signal. A signal flow graph portrays how various signal processing blocks are interconnected. The C++ codes of the blocks in GNU radio are accessed by python codes by importing them using SWIG (Simplified Wrapper and Interface Generator). In the flow graph vertices represent signal processing blocks and edges represent the data flow between them. Flow graphs in CPU can be used along with an USRP to perform radio applications.

IV. IMPLEMENTATION

WaveGuru is a platform to generate and analyse baseband I/Q signals which are compliant to specific technology standards. The WaveGuru box is used as the USRP. It is used to implement GNU Radio. It supports many technologies such as Rel-9 LTE (22 MIMO), WiMAX IEEE802.16e specification, Rel-9 DC-HSDPA and DC-HSUPA , GNU Radio. WaveGuru has two platforms: RF Board and Baseband platform. Some of the RF Board features are as follows: Software tunable across wide frequency range (70 MHz to 6.0 GHz) with a channel bandwidth of 200 kHz to 56MHz. Transmitter and Receiver ports, External Reference Clock source connectivity, ADC, DAC and Modulators, Digital Filters. Some of the features of the Baseband platform are:
512 MB of SDRAM & 16 GB of Flash, 8 KB of I2C EEPROM, Real-Time Clock,10/100/1000Ethernet From frequency deviation, modulation index (mf) can be defined as: Interface, USB-UART Interface.

Two softwares are mainly used in order to accept the real time data from the user. GNU Octave (text) and SOX (voice) are the softwares used .

1. GNU Octave:

GNU Octave is one of the major free alternatives to Matlab. It is primarily intended for numerical computations. Octave helps in solving linear and nonlinear problems numerically, and for performing other numerical experiments using a language that is mostly compatible with Matlab. It can be also referred as a clone of Matlab.The octave code helps in taking the real time text input from the user and is later saved to a file.

2. SOX(Sound Exchange):

SOX is also commonly known as, the Swiss Army knife of sound processing programs. It is a cross-platform (Windows, Linux, MacOS X, etc.) command line utility that can convert various formats of computer audio files in to other formats. It can also apply various effects to these sound files, and, as an added bonus, SoX can play and record audio files on most platforms.

The command `play' is used to play a required audio file from the system. `rec' is used to record voice or audio file. It even gives the user the liberty of choosing the sampling rate, bit depth as well as audio file format.

FM(Frequency Modulation):

Renowned for its high quality performance, frequency modulation (FM) is the most commonly used analog modulation technique, with particular exertion in the VHF band for FM radio systems.FM changes the current frequency of the modulated (transmitting) signal depending on the amplitude of the modulating signal, as is shown in Fig.6.

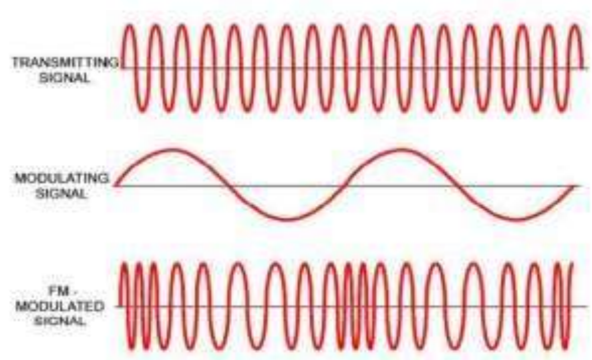

Fig.6: Frequency modulation of the transmitting signal.

As it can be seen, the signal varies as a function of voltage of modulating signal. The amount by which this frequency variation occurs is important, and is known as the frequency deviation and is typically measured in kHz.

$$mf = \frac{\Delta f}{fm} \tag{1}$$

where Δf and fm are the frequency deviation of modulated signal and the highest frequency component of the modulating signal, respectively.

ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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When mf is small (approximately <=0.4), frequency deviations are small as well, and modulated signal consists only of the transmitted signal component and two side components. This is called “NarrowBand Frequency Modulation (NBFM)". When mf is larger, frequency deviations are also larger, and modulated signal, besides transmitted signal component, also contains multiple side components. This is known as “WideBand Frequency Modulation (WBFM)". Although higher quality of transmission can be achieved using WBFM, NBFM uses narrower bandwidth for modulation, accomplishing better spectral efficiency.

Voice Transmission:

a. Transmitter:

Transmission of voice is acheived using Wideband Frequency Modulation(WBFM) and Narrow Band Frequency Modulation(NBFM) techniques. The Fig.7 and Fig.8 are the flow graphs built using WBFM and NBFM respectively.

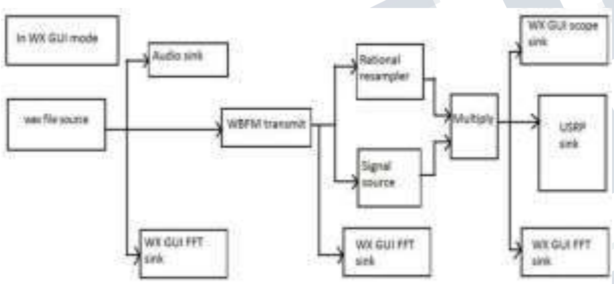

Fig.7: WBFM transmitter.

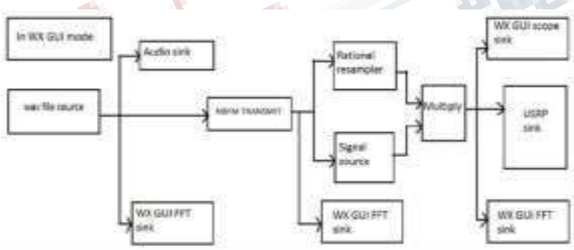

Fig.8: NBFM transmitter.

b. Receiver

The voice transmitted in real time is received using wideband frequency demodulation technique as well as narrow band frequency demodulation technique. As the frequency is modulated, not the amplitude the effect of noise will not hamper the message to be received. The received file is verified by using audio sink. The Fig.9 and Fig.10 represents the WBFM receiver flow graph and NBFM receiver flow graph respectively.

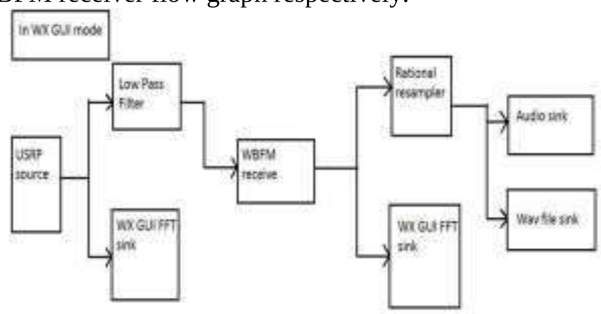

Fig.9: WBFM Receiver.

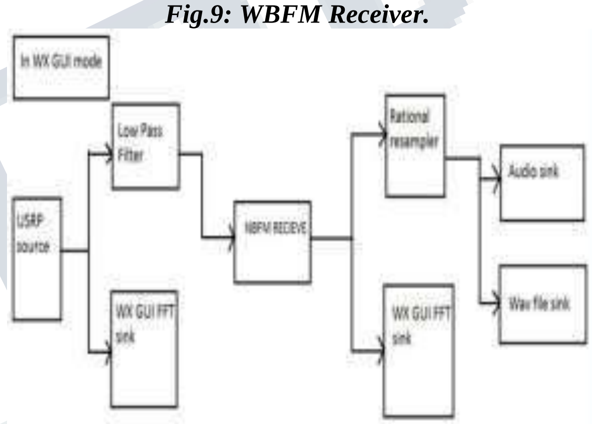

Fig.10: NBFM Receiver.

V. RESULTS

Fig.11 shows the experimental setup where one system acts as transmitter while the other as receiver. FM antennas are used for transmission of voice or audio.

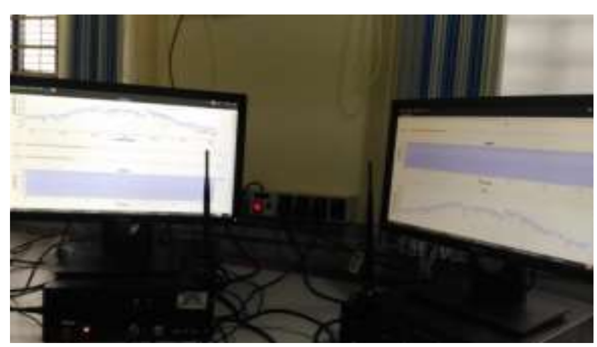

Fig.11: Setup for transmission and reception of audio signal.

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International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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WBFM :

The flow graph of WBFM transmitter and receiver is as shown in Fig.12 and Fig.13 respectively. Voice is recorded using microphone and SOX software. In the SOX, input voice is recorded using the command `rec -c 1 -r 44100 -b 8 record.wav' where, `c' represents the number of channels through which audio is taken as input, `r' is the sampling rate of audio that is being recorded, 44100Hz is a standard sampling rate of any audio file, `b' is the bit depth of the audio, `record.wav' is the audio file stored by the name `record` in a required directory in .wav format. This path is specified in Wav file source. It is listened through the audio sink.

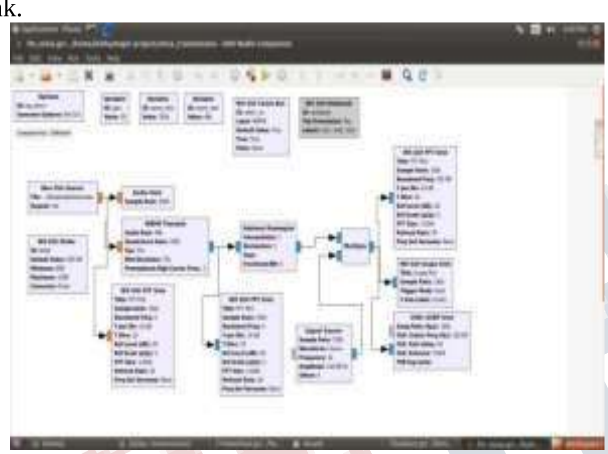

Fig.12: Flow graph of WBFM transmitter.

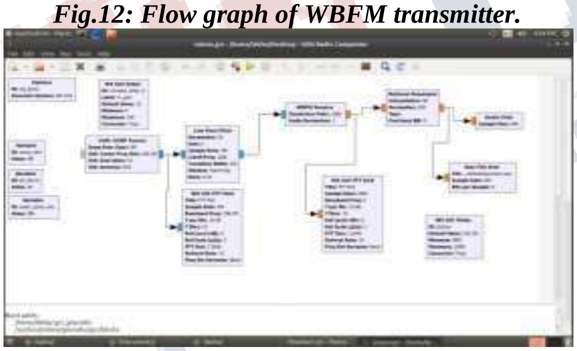

Fig.13: Flow graph of WBFM receiver.

Maximum deviation is the maximum difference between an FM modulated frequency and the nominal carrier frequency. Maximum deviation for WBFM is varied for different line of sight(LOS) and different non line of sight(NLOS) distances to achieve average power level at the receiver. The results are tabulated in the following Table.1 and Table.2.

The ideal frequency deviation for WBFM should be 75K, but the maximum deviation is changed to 150K and 500K for various distances.

Table.1: Performance analysis of WBFM for NLOS distances

SL No.	Antenna Distance	Max Deviation	Rx FFT Average power level
1	6.5m(NLOS)	75K	-89.436db
2	6.5m(NLOS)	150K	-88.065db
3	6.5m(NLOS)	500K	-86.806db
4	10m(NLOS)	75K	-81.794db
5	10m(NLOS)	150K	-81.218db
6	10m(NLOS)	500K	-81.156db

Table.2: Performance analysis of WBFM for LOS Distances

SL No.	Antenna Distance	Max Deviation	Rx FFT Average power level
1	2.72m(LOS)	75K	-82.175db
2	2.72m(LOS)	150K	-81.641db
3	2.72m(LOS)	500K	-81.586db
4	4.42m(LOS)	75K	-86.179db
5	4.42m(LOS)	150K	-85.641db
6	4.42m(LOS)	500K	-85.195db

From the above tables, it can be inferred that, as maximum deviation increases the average power level of reception decreases which results in deterioration of voice quality. It can be also observed that increase in distance also affects the power level of received data.

NBFM:

The NBFM transmitter and receiver flow graphs are shown in Fig.14 and Fig.15 respectively.

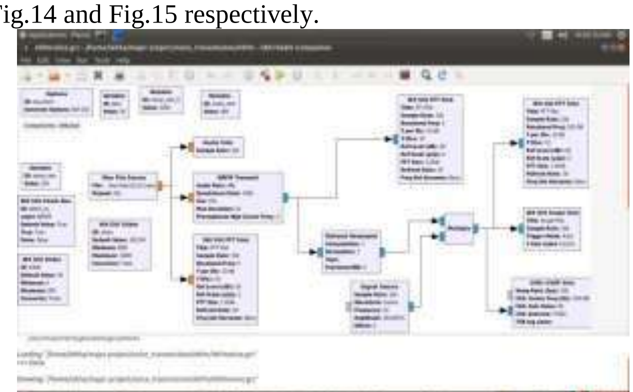

Fig.14: Flow graph of NBFM transmitter.

IFERP
ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
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Ideal maximum deviation for NBFM is 5K, which is varied for different cases and are tabulated in the Table.3. The maximum deviation is varied and the average power level of received voice is affected.

Table.3: Performance analysis of NBFM

SL No.	Antenna Distance	Max Deviation	Rx FFT Average power level
1	6.5m(NLOS)	1K	-92.934db
2	6.5m(NLOS)	2.5K	-92.092db
3	6.5m(NLOS)	5K	-91.977db
4	2.72m(LOS)	1K	-91.174db
5	2.72m(LOS)	2.5K	-91.590db

The received voice quality is graded with grades ranging from 0 to 10(Where grade 0=Pure noise and grade 10=the perfect sound) and tabulated in the Table.4. The grades should not be perceived as absolute, rather they serve as an orientation as to how change in particular parameter affects a perceived quality of received sound. Grading the received quality was based on subjective experience since none of the objective methods were at disposition.Table.5 gives the response to gain variation of NBFM.

Table.4: Response to gain variation of WBFM

SL No.	Tx Gain	Rx sound quality
1	40	3
2	46	5
3	50	5.5
4	62	7
5	87	9

Table.5: Response to gain variation of NBFM

SL No.	Tx Gain	Rx sound quality
1	36	2
2	50	4
3	120	5

energy concentrated in a small bandwidth, resulting in a sharp peak around the center frequency. WBFM, on the other hand, has energy spread over wider spectrum, resulting in a smaller peak.

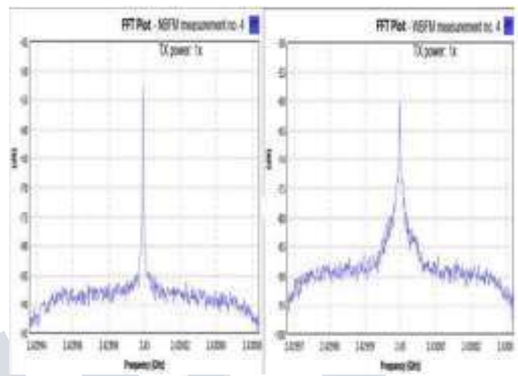

Fig.16: Comparison of NBFM V/s WBFM.

VI. CONCLUSIONS

In this paper, voice is successfully transmitted over wireless media using GNU radio and USRP. A 52MB size of audio book is also successfully transmitted between systems situated at different distances using WBFM and NBFM modulation techniques. With accurate maximum deviation, sampling rate and gain, voice can be received in the receiver system without any noise.

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