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Design and Implementation of Frequency Modulated Transmission and Reception of Speech Signal and FPGA Based Enhancement

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Abstract

Communication system may be the fastest growing technologies in our culture today. One of the ramifications of that growth is a dramatic increase in the number of professions – where an understanding of these technologies is essential for success – and a proportionate increase in the number and types of students taking courses to learn about them. Developments in communication technology have increased its application in allied fields of electronics, including computers and industrial control. Regardless of a student's ultimate area of specialization, knowledge of communications concepts and applications is no longer optional, it is essential to understand today's multidisciplinary applications. This paper provides a basic methodology of designing Transmission and Reception parts of a speech signal using different electronic equipment's which can be easily understandable. Then we have applied here a speech enhancement filtering technique using FPGA and finally it is played via headphone.

Keywords

System Generator, AC'97 CODEC, FIR Filter, JTAG Co-sim, Simulink.

1. Introduction

Communication simply means the process of sending messages or information from one place to another. In the past, communication was done using signs, fire, drum beats, runners and even carrier pigeons [19]. In modern times, these crude ways of communication have been replaced by electrical communication in which electrical signals are used for communication[1].

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Now communication has evolved into telegraphy, telephony, radio, microwave, satellite, mobile, optical and computer communications. Earlier electrical communication was analog in nature[4]. The earliest analog communication system namely line-telegraphy originated in 1840[3]. In recent years communication has become more widespread with the use of satellites and fibre optics[2]. But analog communication system is now gradually replaced by digital communication system for the main advantages of reducing noise effect in channel and the encryption scheme is easier in digital communication system than in analog communication system. In 1937, pulse code modulation which provided a means for converting analog signals to digital signals for digital coding of speech signals, was invented [7]. This paper presents a basic knowledge on a simple communication system through which a voice or speech signal will be transmitted and received and finally enhanced by a filtering technique which is used to remove a certain range of unwanted frequencies. The transmission and reception process is done by Frequency Modulation technique for which we have used separate analog circuits. After reception we have inputted the audio signal to a FPGA board named Virtex ML506 and there it is digitally filtered. Finally we have heard the audio by connecting the headphone to the Line-out jack of Virtex board. We have simulated the proposed filtering design both in simulink and Xilinx System Generator and finally implemented onto the specified FPGA board.

There are so any related works on this. D.K.Sharma, A.Mishra and Rajiv Saxena has been described a brief overview over different analog and digital modulation techniques through extensive literature survey in a tabular manner enabling to analyze and establish the superiority at a glance of a specific modulation technique for a particular application in an International Journal Paper[23]. Murat Tanyel, Kathrine Nguru implemented a virtual tool kit for communication system and also presents some examples of the virtual instruments (VIs) developed in their paper. Shahnam Mirzaei, Anup Hosangadi, Ryan Kastner presented a method for implementing high speed Finite Impulse Response (FIR) filters using just registered adders and hardwired shifts. They extensively used a modified common subexpression elimination algorithm to reduce the number of adders. They targeted their optimizations to Xilinx Virtex II devices where their compared their implementations with those produced by Xilinx CoregenTM using Distributed Arithmetic [24]. Apurva Singh Chauhan and Vipul Soni have designed a FIR filter using FPGA using IP-Core. The filter is set to 16-bit signed data processing. IP Corse has been used to filter the input data. The design is coded through VHDL (hardware descriptive language). To verify the designed outputs simulation, compilation and synthesis have been done [21] [25].

2. Background Theory

Analog communication is that type of communication in which the message or informational signal to be transmitted is analog in nature. This means that in analog communication the modulating signal is analog such as speech, video shooting etc [5].

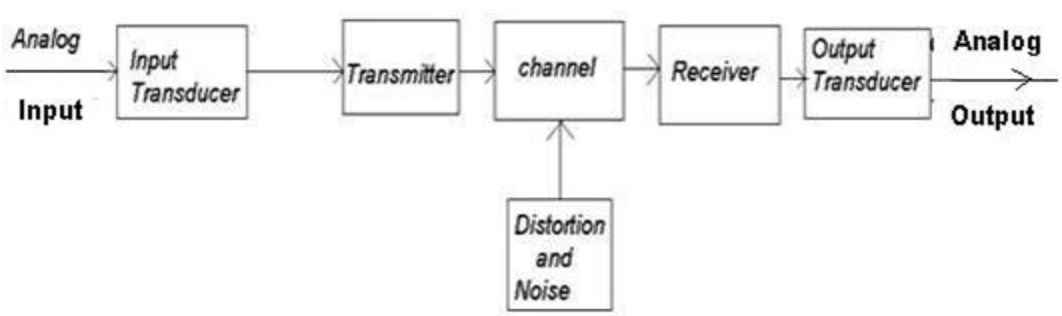

Figure 1: Analog Communication System

Presently all AM, FM radio transmission are examples of analog communication [8].

Table 1: Performance analysis of analog modulation scheme

Sr. No.	TYPE OF ANALOG MODULATION	BANDWIDTH (B. W.)	% POWER SAVING	POWER REQUIREMENT
1	AM-DSB-FC	$2\omega_m$	Standard	3/2 Pc
2	AM-DSB-SC	$2\omega_m$	66.67%	5/4 Pc
3	AM-SSB-FC	ω_m	16.67%	1/2 Pc
4	AM-SSB-SC	ω_m	83.33%	1/4 Pc
5	AM-VSB	$> \omega_m$	>SSB-SC	Greater than SSB-SC
6	NBFM	$2\omega_m$	Same as DSB-SC	Same as DSB-SC
7	WBFM	$2\omega_m M_F$	More than NBFM	More than NBFM
	ω_m = modulating frequency	M_F = modulation index in FM	M_P = modulation in PM	Pc = carrier power

We here use the frequency modulation scheme because it has certain advantages:

i. FM receivers are more immune to noise.
ii. It is possible to reduce noise still further by increasing the frequency-deviation.
iii. Standard frequency allocations provide a guard band between commercial FM stations.
iv. FM broadcasts operate in the upper VHF and UHF frequency ranges at which there happens less noise than in the MF and HF ranges.
v. The amplitude of FM wave is constant, thus independent of modulation depth that governs the transmitted power. This permits the low level modulation in FM transmitter and use of class C amplifiers in all stages following the modulator.

This technique is mostly used in FM Radio, FM Broadcasting, Magnetic Tape storage etc.

2.1 Transducer

The input quantity for most instrumentation system is a "non-electrical quantity". In order to use electrical methods & techniques for measurement, manipulation or control, the non-electrical quantity is converted into an electrical signal by a device called a "transducer (mostly referred to as Electric

Transducer)" [6]. The transducer may be thought of consisting of two important and closely related parts. These two parts are: **Sensing element** which responds to a physical phenomenon or a change in a physical phenomenon and **Transduction Element** which transforms the output of a sensing element to an electrical output[9]. Here we have used a carbon microphone.

2.2 FM Transmitter

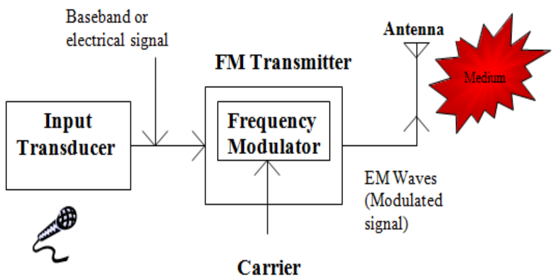

Figure 2: FM Transmission block

Here first of all a voice signal is inserted into the system using a microphone. This micro phone which acts as a Transducer, converts the voice signal into electrical signal [22]. This electrical signal is then applied to FM transmitter where the signal is frequency modulated with a carrier frequency. This modulated signal is then transmitted to the medium through a transmitting antenna [11][14].

2.2.1 Frequency Modulator

Frequency Modulation FM) is that type of angle modulation in which the instantaneous frequency is varied in linearly with a message or baseband signal about an un modulated carrier frequency [10]. This means that the instantaneous value of the angular frequency will be equal to the carrier frequency plus a time-varying component proportional to the baseband signal. Edwin Howard Armstrong (1890–1954) was an American electrical engineer and inventor who invented frequency modulation (FM) radio[15].

Suppose the baseband data signal (the message) to be transmitted is-

$$x_m(t)$$

and is restricted in amplitude to be-

$$|x_m(t)| \le 1,$$

and the sinusoidal carrier is-

$$x_c(t) = A_c \cos(2\pi f_c t)$$

Where f_c is the carrier's base frequency and A_c is the carrier's amplitude. The modulator combines the carrier with the baseband data signal to get the transmitted signal,

$$\begin{aligned} y(t) &= A_c \cos\left(2\pi \int_0^t f(\tau) d\tau\right) \\ &= A_c \cos\left(2\pi \int_0^t \left[f_c + f_\Delta x_m(\tau)\right] d\tau\right) \\ &= A_c \cos\left(2\pi f_c t + 2\pi f_\Delta \int_0^t x_m(\tau) d\tau\right) \quad ....(1) \end{aligned}$$

In this equation, $f(\tau)$is the Instantaneous frequency of the oscillator and $f_\Delta$is the Frequency Deviation, which represents the maximum shift away from f_c in one direction, assuming $x_m(t)$ is limited to the range ±1[13].

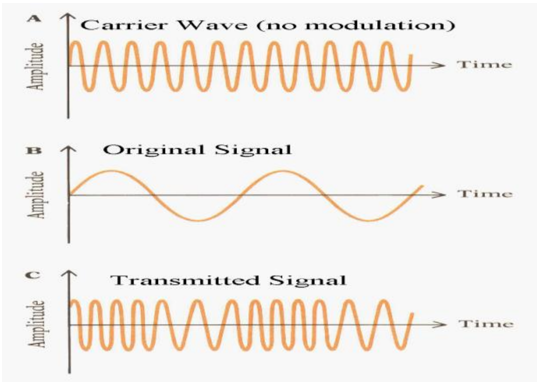

Figure 3: Schematic Diagrams of Carrier, Baseband & Frequency Modulated signals

FM signals may be put into two different categories as under [12].

i. The Direct method or Parameter variation method
ii. The Indirect method or the Armstrong method.

2.3 FM Receiver

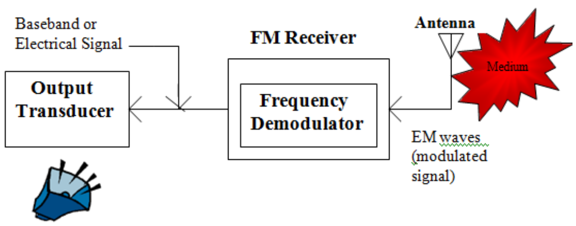

Figure 4: FM Reception block

Here the modulated signal is firstly received through the receiving antenna from the medium and then applied to the FM receiver where the signal frequency demodulated. Then the original electrical signal is inverse transduced by the output transducer like amplifier box or mike etc. which converts the electrical signal into sound or it can be further processed.

2.3.1 FM Demodulator

The FM demodulator perform the extraction of modulating signal in two steps as follows:

i. It converts the FM signal into a corresponding amplitude modulated signal with the help of frequency dependent circuits like L-C circuits. These circuits are generally known as **Frequency Discriminators** [1][16].

ii. The original baseband signal is recovered from this AM signal with the help of the linear diode detector or envelope detector.

FM demodulators are of following types:

i. Slope Detector
- Single-tuned detector circuit or simple slope detector
- Stagger-tuned detector circuit or balanced slope detector.

ii. Phase Difference Detectors
- Foster-Seeley detector
- Ratio detector
- PLL-FM demodulator

2.4. AC'97 CODEC

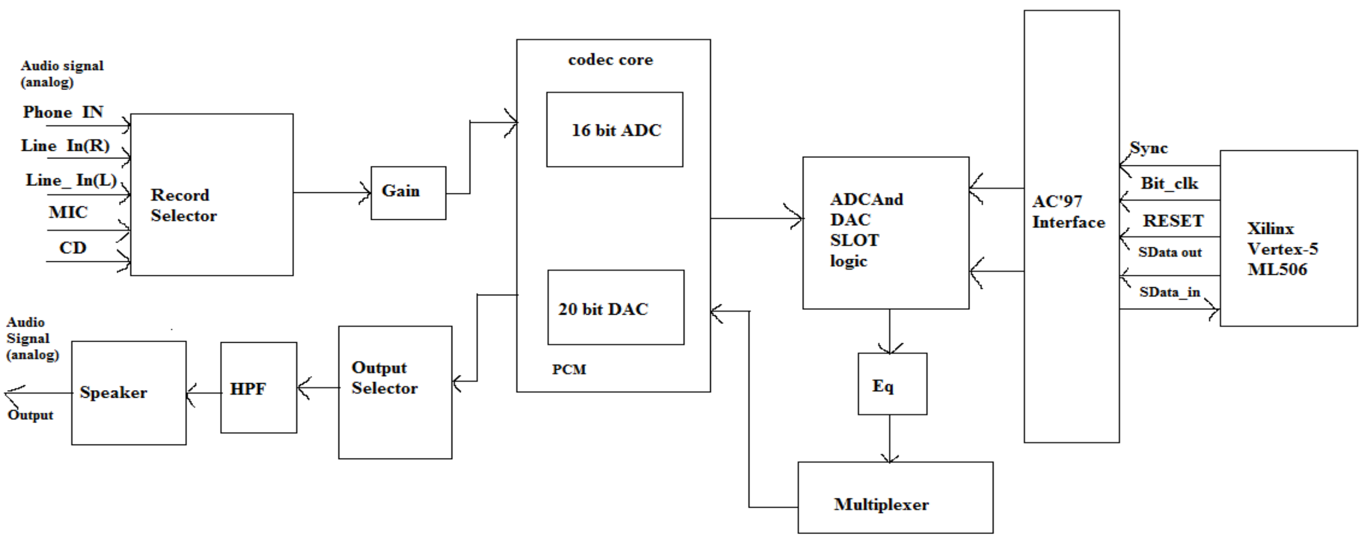

Figure 5: Block Diagram of AC'97 CODEC

The Audio Codec '97 (AC '97) Architecture and Digital Interface (AC-link) specifically is designed for implementing audio and modem I/O functionality in mainstream PC systems. As we are using here on board stereo AC97 audio codec so we have to create HDL wrapper which includes common board support packages (.ngc,ILA and ICON), and Instantiates the AC97 controller, DCM, and System Generator audio design.HDL wrapper Interfaces SysGen design to External Components and is generated from SysGen .mdl model.

2.5. Digital Finite Impulse Response Filter

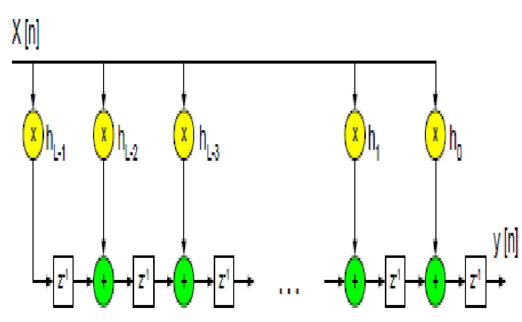

Figure 6: Block Diagram of Digital FIR Filter

Digital FIR filter enhances the signal by performing mathematical operation on the signal [24][25]. The equation of N^{th} order digital FIR Filter is written as under-

$y[n] = \Sigma\, h[k]\, x[n-k]$……….(2)

In equation 2, N is filter order, y(n) is the filtered signal h(n) is the coefficient of a sample and x(n-k) is the input signal. It is used here to remove the unwanted range of frequencies [18]. Digital FIR filter performance can enhanced by varying filter order and cut off frequency. There are several tests performed to choose filter order and cut off frequency for better stereo sound effect [17].

So the total block diagram becomes like the following figure –

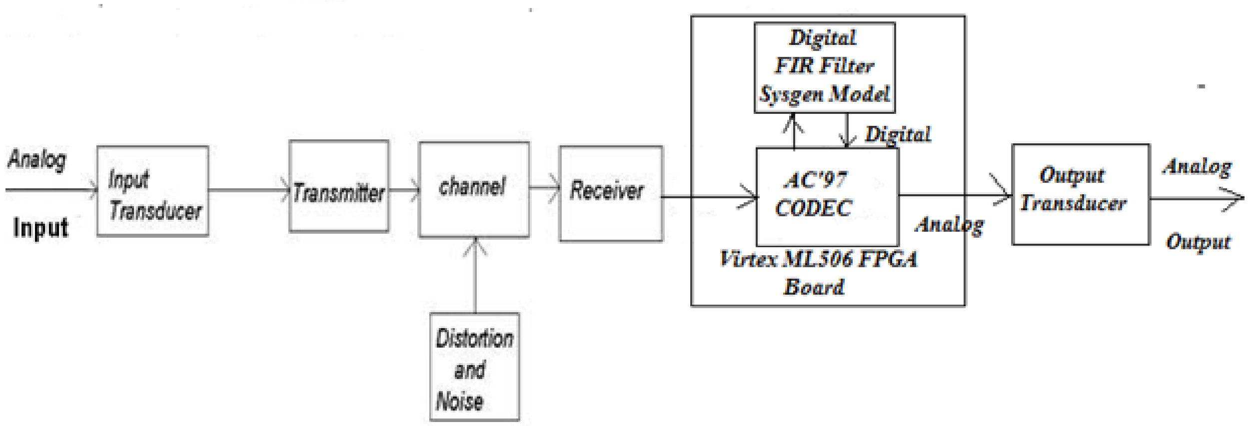

Figure 7: System block diagram

Here the speech signal is inputted to the transmitting block through an input transducer. After FM modulation it transmitted through an antenna. We have used wires for antenna purpose. After receiving the signal it is FM demodulated and again inputted to the FPGA Virtex board where it is digitally enhanced by software based filtering technique and finally get outputted by an output transducer.

3. Proposed Architecture Scheme

3.1. FM Transmitter Circuit

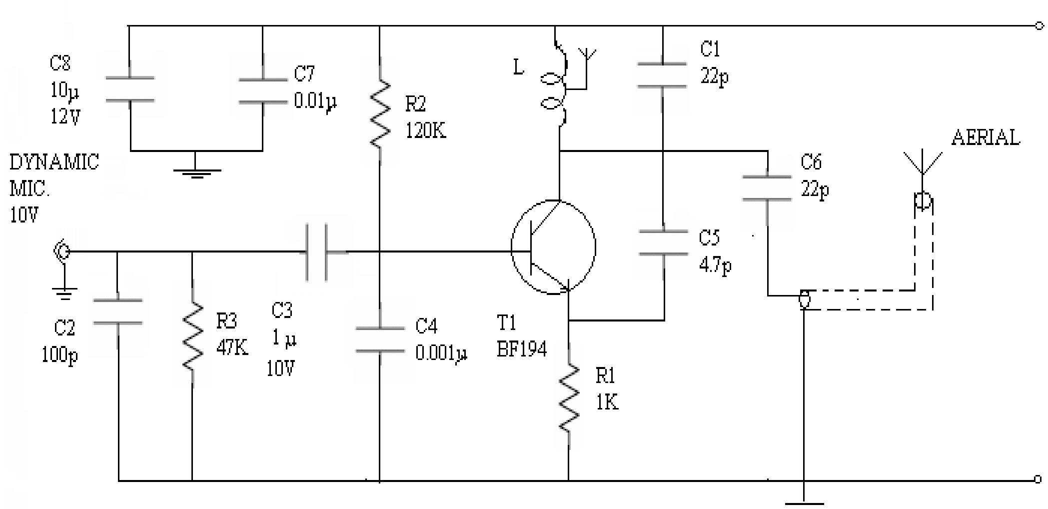

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Figure 8: Transmitter circuit diagram

3.2. FM Receiver Circuit

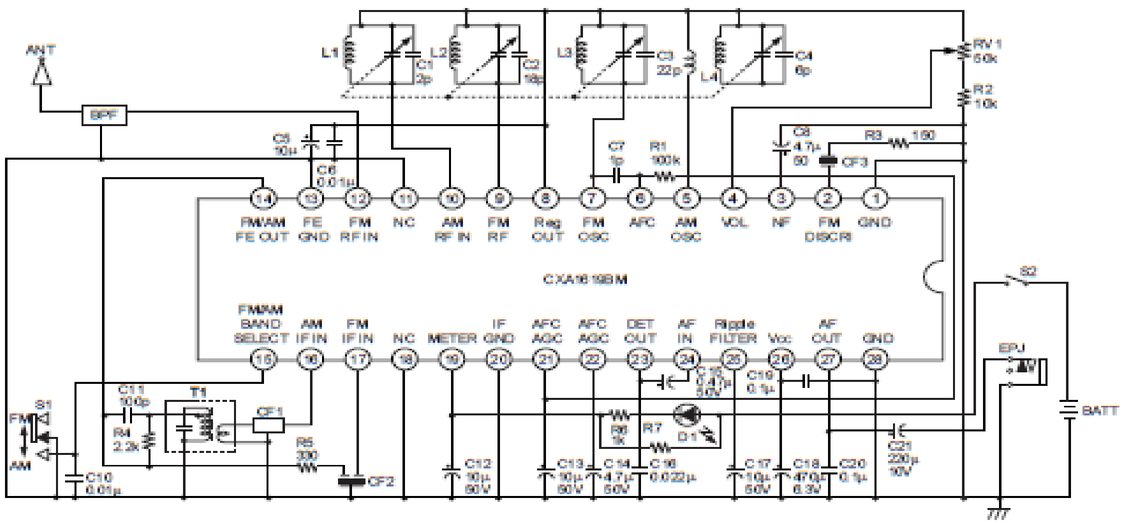

Figure 9: Receiver circuit diagram

Here we have used an IC named CXA1619BM which is a bipolar mono-lithic one-chip FM-AM radio IC designed for radio-cassette tape recorders and headphone tape recorders and has small number of peripheral components, low current consumption(V_{cc} = 3V, I_D= 5.8mA), Built-in FM/Am Select Switch, large output of AF Amplifier etc.

3.3. System Generator Block of Digital FIR Filter

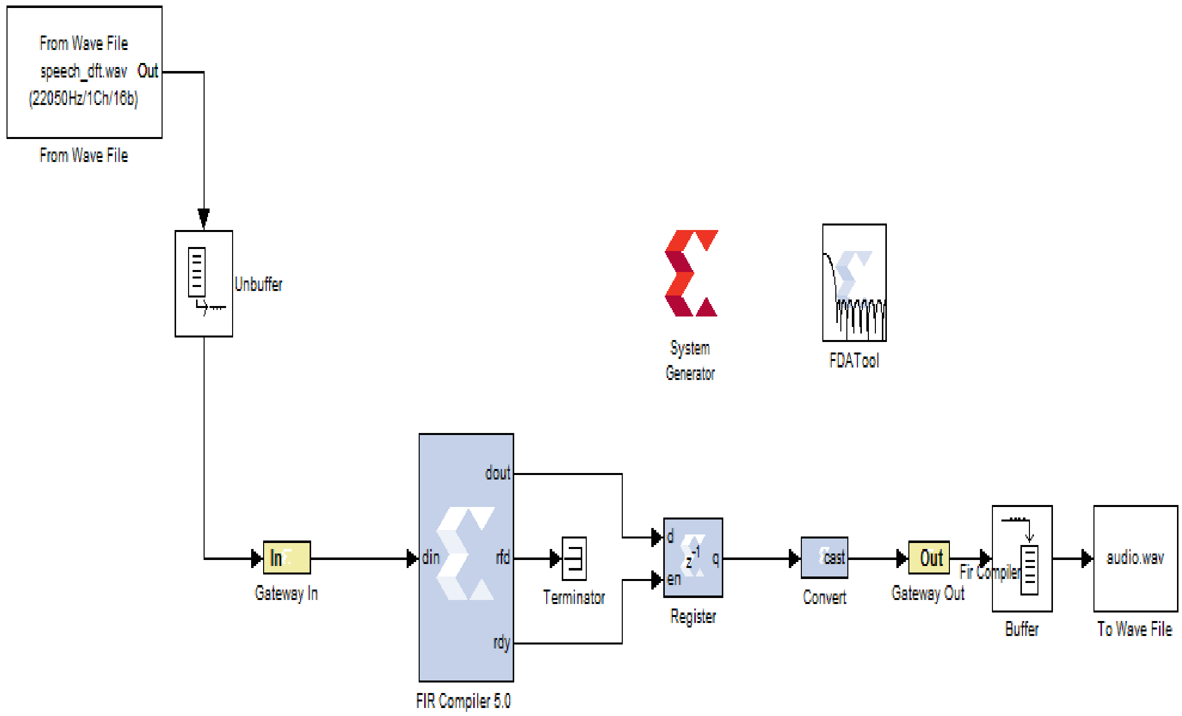

Figure 10: System Generator Model of Digital FIR Filter

4. Software Implementation

We here used Xilinx System Generator Software which allows device-specific hardware designs to be constructed directly in a flexible high-level system modelling environment. In a System Generator design, signals are not just bits. They can be signed and unsigned fixed-point numbers and changes to the design automatically translate into appropriate changes in signal types. Blocks are not just stand-ins for hardware. They respond to their surroundings, automatically adjusting the results they produce and the hardware they become. The simulation results are bit and cycle-accurate.

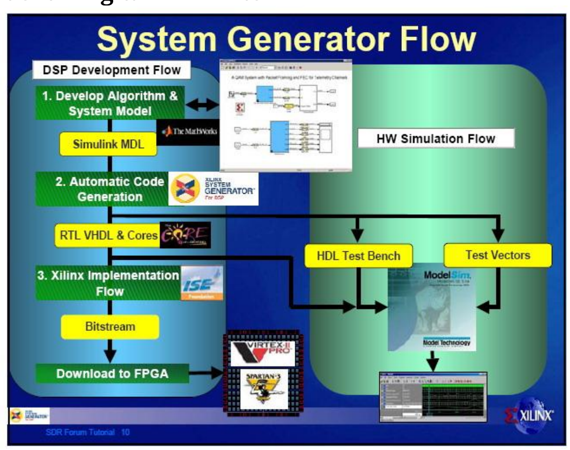

Figure11: Typical Design Flow for System Generator Software

5. Hardware Implementation

5.1. JTAG Co-simulation Block

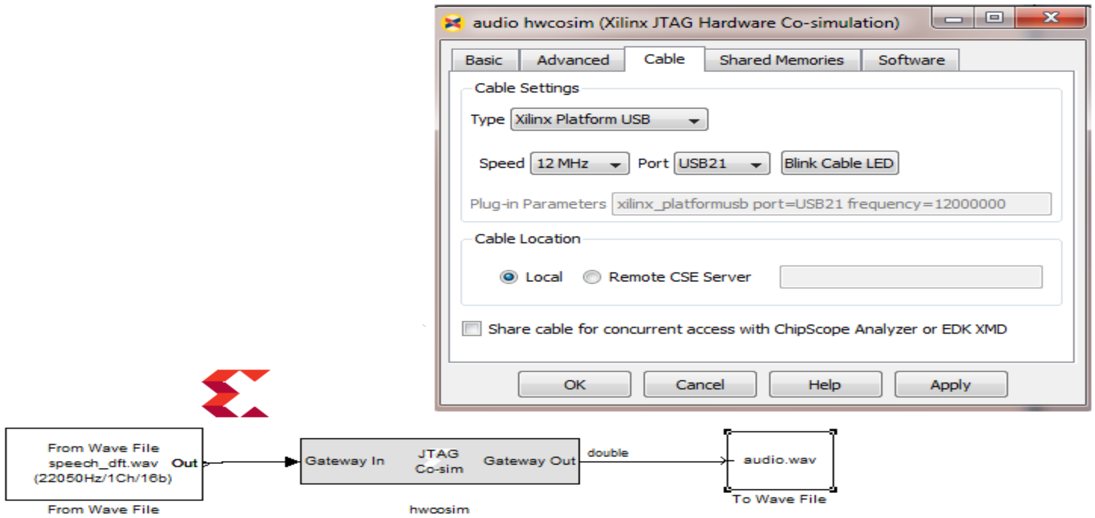

Figure 12: Hardware/Software Co-simulation

System Generator provides several methods to transform the models built using Simulink into hardware. One of these methods is called Hardware/Software co-simulation. Hardware/Software co-simulation enables building a hardware version of the model and using the exile simulation environment of Simulink we can perform several tests to verify the functionality of the system in hardware. HW/SW Co-simulation supports FPGAs from Xilinx on boards that support JTAG or Ethernet connectivity.

5.2. Virtex ML506 FPGA Board

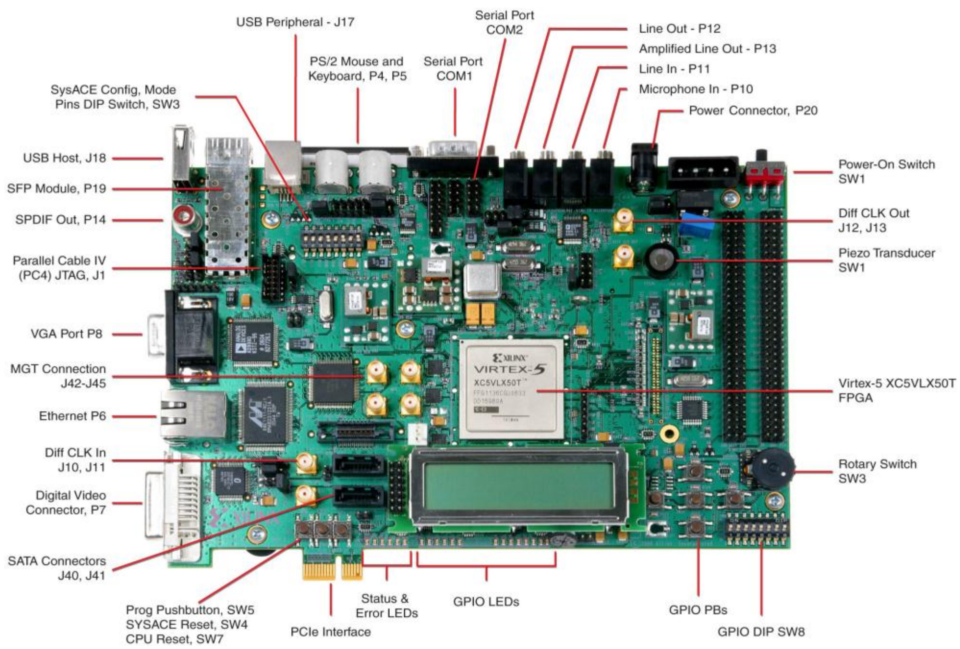

Figure 13: Xilinx Virtex ML506 board

The XtremeDSP™ development platform—Virtex®-5 ML506 FPGA edition is a feature-rich DSP general purpose evaluation and development platform. Though economically priced, the ML506 offers users the ability to create DSP based and high speed serial designs utilizing the Virtex-5 FPGA DSP48E slices and RocketIO™ GTP transceivers. A variety of on-board memories and industry standard connectivity interfaces add to the ML506's ability to serve as a versatile development platform for embedded applications. But the basic cause of choosing this board is it has on-board AC'97 CODEC which is very much essential for our speech signal processing [20].

6. Results And Performance Analysis

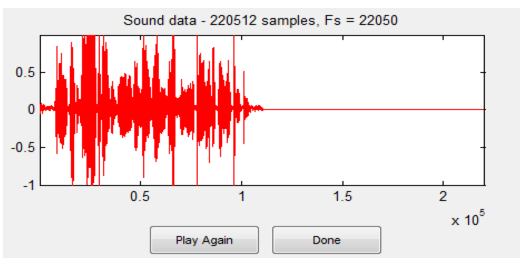

Figure 14: Input signal spectrum from FM Receiver

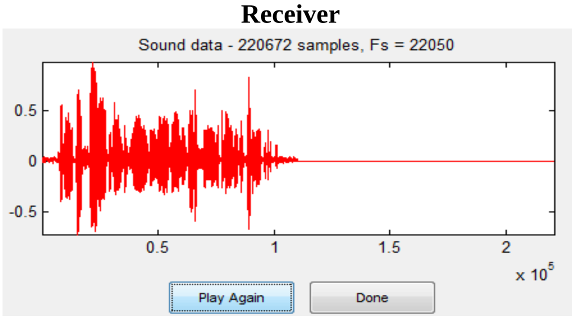

Figure 15: Output signal spectrum Digital Filter block

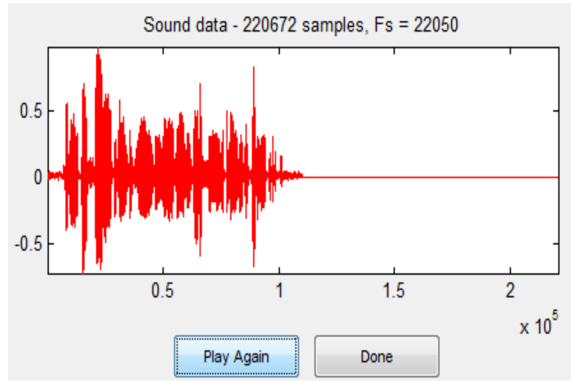

Figure 16: Output Spectrum of Hardware In Loop JTAG Co-Simulation

Here we can see that the sound coming from the receiver sounds a bit noisy. Clear sound is not coming. So we apply the filtering technique and get the cleared and sharp output of that voice signal.

7. Conclusion And Future Work

In our design we used a simple analog communication system. But there is some drawbacks. So now Digital Communication System is widely used. The reasons behind choosing digital are:

1) The digital communication system is simpler & cheaper compared to analog communication system because of the advances made in the IC technologies. Though the digital communication system comprises more equipment's than analog. But we have to fabricate all the components used in analog. So it is very time consuming & costly. But the components used for digital are available in the market.

2) The speech, video & other data may be merged & transmitted over a common channel using multiplexing very easily.
3) Using data encryption, only permitted receivers may be allowed to detect the transmitted data. Only digital communication system supports this. This property is of its most importance in military applications.
4) Since the transmitted signal is digital; the channel encoding is used so the noise does not accumulate from repeater to repeater in long distance communication. As

the signal is digital a large amount of noise interference may be tolerated.

5) Error detection & correction capability is higher in digital because though the binary digits gets corrupted by noise but by proper error detection & correction methods we can recover the original bits as well as possible. This is done by encoder & decoder. Last of all digital communication system is adaptive to other branches of data processing.

So we will apply here a digital transmission system in future and also some other FPGA based enhancement techniques like time-stretching, reverberation, echo, equalization etc. We hope that the report will provide the basic knowledge of designing a simple communication system to the interested people and welcome any suggestions towards the improvement of this project.

Acknowledgment

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