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***Abstract*—As communication plays an important role in day to day life, the effective and efficient voice transmission is to be maintained. This paper mainly deals with voice transmission over a channel and implemented using 6713 DSK. For this purpose, some modulation schemes and voice coders are implemented. So two points of view are developed. First, a static point of view, using a prototype on MATLAB, estimates the different combinations' performances, using a stored speech sample. Then, a more dynamic point of view tests the system in real time, using C code adapted from MATLAB and embedded on DSPs, with the actual transmission channel being emulated by another DSP. In MATLAB the voice signal using different techniques are simulated and the outputs for modulation and demodulation signal are obtained which are shown in this paper for random bits operation of signals. An optimal transmission/reception scheme intended for voice transmission on DSP Processor TMS320C6713 is done using hardware and the results are compared in MATLAB by maintaining proper accuracy.**

Keywords: Modem, TMS320C6713, Voice Transmission, VoIP Matlab.

I. INTRODUCTION

A typical communication system will input data, perform some form of processing and frequency translation, transmit the data, and perform the converse operations at the receiver. The transmission of data is done using Public Switched Telephone Network (PSTN). The modem contraction of the term modulator-demodulator is a conversion device that facilities the transmission and reception of data over a public switched telephone network (PSTN).

Modem:

The Modem is a contraction of the term modulator, demodulator, is a conversion device that facilities the transmission and reception of data over PSTN [1]. The analog signal is converted into digital format and is passed through modulator and demodulator reconverts the digital format signal and channel output.

This paper deals with voice transmission over a channel. For this purpose, some modulation schemes and voice coders are to be implemented. Voice coding and transmission is an important and pervasive task in any telecommunication system. A simple transmission/reception scheme intended for voice transmission is proposed in the figure below consisting of: one microphone, a DSP board for voice acquisition, sampling, coding and modulation, a channel emulator, a second DSP board for voice processing or signal decoding, demodulation and voice reconstruction, a loudspeaker, and finally two PCs that control each DSP[15].

II. TRANSMITTER SECTION

Voice coding and transmission is an important and pervasive task in any telecommunication system. Actually, voice transmission has been spread out from the classical telephone oriented network (POTS) to the packet oriented networks like the Internet (VoIP)[4].

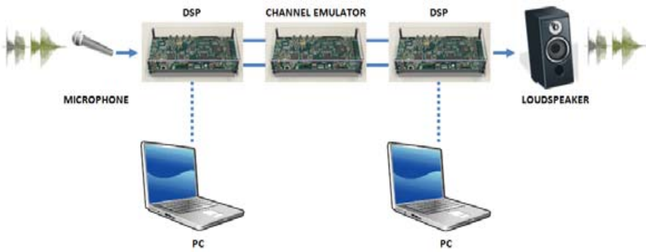

Figure1a.

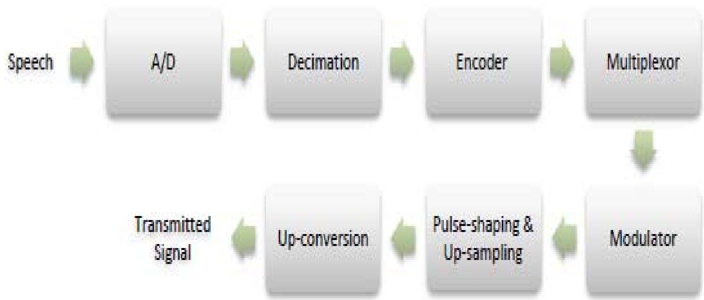

Figure1b

Fig1. Transmitter Section

The input of the transmitter is the recorded speech by the microphone. An A/D converter samples the voice to obtain digital data from this input signal.

The data is then down- sampled and encoded. Finally up conversion is used to transmit through the physical channel.

III. Receiver Section

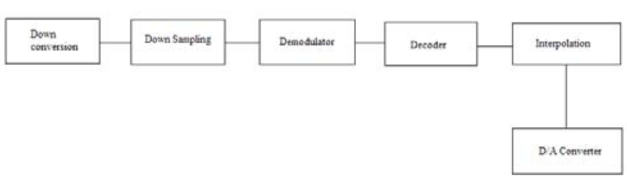

Fig2.Receiver Section

The microphone records the speech data to be transmitted; while the main purposes of the transmitter are to encode the speech to minimize its bit rate, and modulate the data in such a way as to produce electromagnetic signals that can be transmitted across the channel. The channel transmits the signal. Regarding the receiver, the output of the channel is first synchronized, then the influence of the channel over the signal is estimated to correct the errors of the received data. Finally, the data is demodulated and decoded, and the reconstructed speech displayed by a loudspeaker.

By using the modulation techniques of BPSK and QPSK the matlab outputs of these modulation techniques are also determined.

The sequence of BPSK coding technique explains clearly the BPSK is a digital modulation technique that separates bits by shifting the carrier180 degrees [3]. A carrier frequency signal is chosen that is known by both the transmitter and the receiver. Each bit is encoded as a phase shift in the carrier at some predetermined period. When a 0 is sent, the carrier is transmitted with no phase shift, and when a 1 is sent, the carrier is phase-shifted by 180 degrees [2, 3].

In the digital information where the speech in digital encoded form with a bit rate of 16 kbit/s, and after Forward Error correction (FEC) coding the data rate becomes 32 kbits/s. we can keep the channel about 1 GHz, range which is better to transmit the 32 kb/s[5]. if the phase of the carrier is switched between 32kbits/s, being at 0 deg and 180 deg for bits having logic 0 and 1 then there are 31,250 radio frequency oscillations per bit transmitted. The output waveform shown below for the logic 0 and 1 is shown below for the 0 deg and 180 deg.

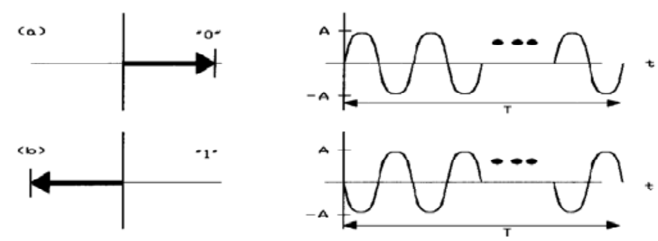

Figure3: BPSK output waveform

For logical 0 the phasor diagram is shown along the positive axis and logical 1 the phasor diagram is shown along the negative axis. For the sequence of the bits the modulator output of the BPSK is shown below, which is the combination of the above diagram:

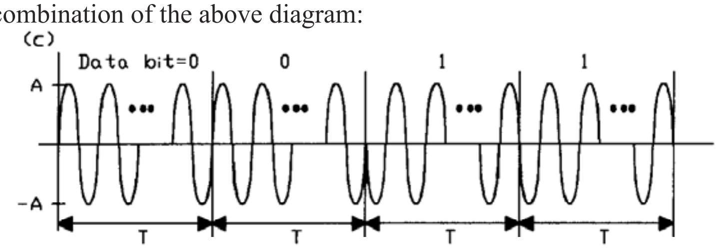

Figure4: BPSK waveform II

BPSK modulation technique the constellation diagram shown below which are transmitting binary data there are only two points. As these two points are at equal distance from origin we would expect them to represent equal magnitude carriers.

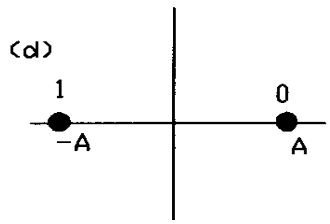

Figure5: BPSK Constellation Diagram

In Binary Phase shift keying (BPSK), determining the phase of transmitted signal is decoding the transmitted information[10].This problem is also known as the carrier recovery attempt made to recover the phase of the carrier. When the phase point at say,180deg is selected to reflect the information being transmitted , the phase of the transmited carrier is set to 180deg.

The Implementation of BPSK Modulation and Demodulation Technique for a random bit stream where "Signal to Noise" Ratio, Signal Power is initialized. We generate the "Noise Power" and "Noise Standard Deviation"[16].

a = E_s^2/SNR; …….(1)

sig_n = sqrt(No/2); ……. (2)

For the Standard Values of "Frequency, Time Vector" and Number of Bits are used for generation of BPSK signal. We generate a Signal Vectors with the Signal Power and Cosine Function.

S (ii, :) = E_s*cos(w*t-2*pi*ii/2)……(3)

Modulate the random bit stream and then plot the transmitted signal and create the Noise Vector join both the signals and generate Received signal. We will generate the basic functions of Cosine and sine functions and detect the signal at output. For the output obtained we check the decision on received bits[11].

The BPSK signal generated using .Wav File function where the Wav file converted in BPSK signal format there by the signal is converted into binary format where XOR function is applied and the signal is generated in binary form.

The transmission of an encoded BPSK signal with voice as input and the reception (demodulation) of this signal with phase-locked loop (PLL) support on a second DSK. In BPSK, the receiver must be able to lock onto the phase of a received signal in order to distinguish between 1s and 0s where PLL is used to lock the information and generate the signal. A sinusoid of 1 kHz, with varying phase, is used as the real-time input to the DSK [4].This input signal has eight unique phase shifts. The real-time output signal is the phase of the received signal. Two DSKs are required to implement this project in Real-Time DSK[16].

IV. SIMULATION RESULTS

Simulations are carried out on implementation different techniques and obtained results for them in mat lab and Code Composer Studio (CCS).

In matlab simulation are done for different techniques those are Binary Phase shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK). For each technique outputs are shown above. Quadrature Phase shift keying (QPSK) technique simulation which are shown by taking digital bits and obtain the outputs for them and the same time for random bits also simulation has been done.

The implementation of Binary Phase Shift Keying (BPSK) technique is also done. This technique is also implemented in the C-Code which is done in Code Composer Studio (CCS) and it is implemented in DSK kit and obtained the output [4]. Finally, I conclude that we had obtained the required outputs in both matlab and Code Composer Studio (CCS).

Results for BPSK modulation/demodulation technique for random bit stream in MATLAB are also shown. The plot shows the transmitted and received waveform of the BPSK technique. The Blue color signal shows the transmitted signal and Red color signal shows the received signal.

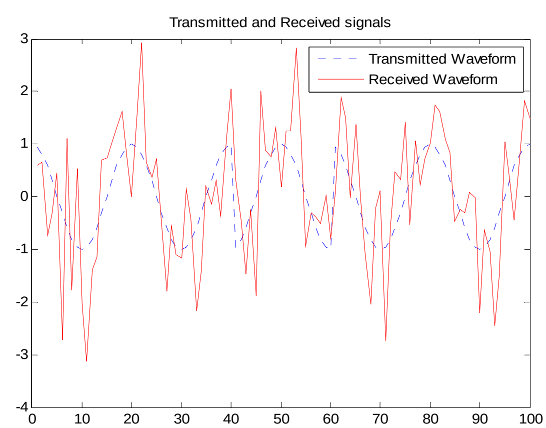

Figure6: BPSK Output waveform

This plot shows the false and error detection region of the BPSK technique which is present. In the region the blue dots shows there are no errors in the region which means that the above obtained BPSK output waveform is correct.

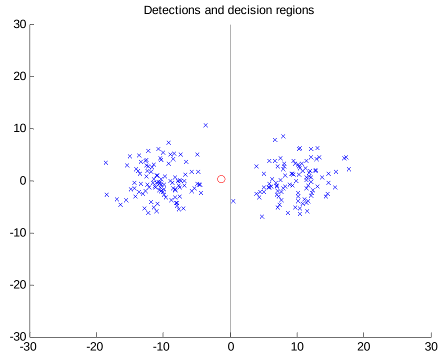

Figure 7: False and Error correction detection

Results for BPSK modulation technique for random bit stream in MATLAB for .Wav file. This plot shows the received waveform and BPSK signal waveform in Amplitude versus Sample. In the received waveform graph it shows the variation in Amplitude with change in sample value .

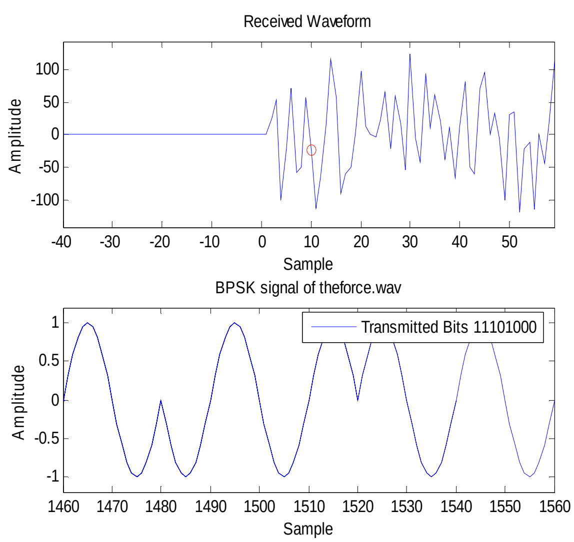

Figure 8: BPSK output waveform for Wav file

V. CONCLUSION

In this the entire work is done by using different techniques in matlab and Code Composer Studio (CCS). In matlab we had implemented Binary Phase shift Keying (BPSK) modulation technique and is also done in a DSP platform using the C6713 floating point DSP. All the basic building blocks of the ADSL modem functionalities were implemented tested and debugged using Matlab. A complete agreement between the transmitted and received data was obtained indicating the success of the implementation.

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