An Efficient Design and Implementation of Software Radio System

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***Abstract* - A radio receiver is a device that picks up the desired signal from the numerous signals propagating at that time through the atmosphere, amplifies the desired signal to the required level, recovers from it original modulating signal and eventually displays it in the desired manner. This outline of functions that must be performed shows that the major difference between receivers of various types is in the way in which they demodulate the received signal and this in turn will depend on the type of modulation employed at the transmitter. The second major difference is the method of displaying the received signals. In this paper, Software Radio System was designed, implemented and the effects of potential channel disturbances were analyzed and compensated.**

***Keywords* – Software Radio System, Signals, Modulations.**

I. INTRODUCTION

A Software Radio system is a transmitter and receiver system that uses digital signal processing for coding, decoding, modulating and demodulation the data. This paper focus on using the IEEE 802.11a specification to create a software radio. The feasibility of using Math works' Simulink and Texas Instrument's Code Composer Studio to design, test, and prototype an OFDM software radio system on a Texas Instruments CDSK6713 DSP development board was studied. Among the subjects examined were communication with the board through real time data exchange (RTDX), quadrature amplitude modulation (QAM), orthogonal frequency division multiplexing (OFDM), frame and carrier synchronization, and issues with Simulink DSP code generation for prototyping.

The basic arrangement of the radio receiver used an antenna feeding an amplifier and down-converter feeding an automatic gain control, which fed an analog to digital converter that was on a computer VME bus with a lot of digital signal processors . The transmitter had digital to analog converters on the PCI bus feeding an up converter that led to a power amplifier and antenna. The very wide frequency range was divided into a few sub-bands with different analog radio technologies feeding the same analog to digital converters. This has since become a standard design scheme for wide band software radios. The goals were to get a more quickly reconfigurable architecture , in open software architecture, with cross-channel connectivity. The secondary goals were to make it smaller, weigh less and cheaper. The project produced a demonstration radio only fifteen months into a three-year research project. The demonstration was so successful that further development was halted, and the radio went into production with only a 4 MHz to 400 MHz range. The software architecture identified standard interfaces for different modules of the radio: "radio frequency control" to manage the analog parts of the radio, "modem control" managed resources for modulation and demodulation schemes (FM, AM, SSB, QAM, etc), "waveform processing" modules actually performed the modem functions, "key processing" and "cryptographic processing" managed the cryptographic functions, a "multimedia" module did voice processing, a "human interface" provided local or remote controls, there was a "routing" module for network services, and a "control" module to keep it all straight. More recently, the GNU Radio using primarily the Universal Software Radio Peripheral (USRP) uses a USB 2.0 interface, an FPGA, and a high-speed set of analog-to-digital and digital-to-analog converters, combined with reconfigurable free software. Its sampling and synthesis bandwidth is a thousand times that of PC sound cards, which enables an entirely new set of applications. In addition the HPSDR (High Performance Software Defined Radio) project uses a 16bit 135MSPS analog-to-digital converter that provides performance over the range 0 to 55MHz comparable to that of a conventional analogue HF radio. The receiver will also operate in the VHF and UHF range using either mixer image or alias responses. Interface to a PC is provided by a USB

2.0 interfaces. The project is modular and comprises a back plane onto which other boards plug in. This allows experimentation with new techniques and devices without the need to replace the entire set of boards. An exciter provides 1/2W of RF over the same range or into the VHF and UHF range using image or alias outputs.

II. RELATED WORK

A. Type I: Hardware model

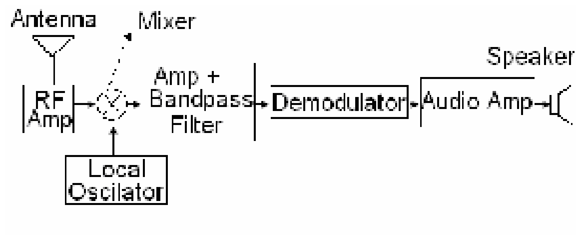

Fig. 1. Analog Receiver Block Diagram.

This SDR must even find solutions to channel and user entire schema from antenna to speaker. As in Fig.1The RF frequency entered in the antenna will be first amplified and filtered to the bandwidth that must be selected. The local oscillator, LO, will down convert the RF signal into new intermediate frequency, IF, that can be treated, filtered and amplified as necessary. Finally, when the expected IF is found, it can be demodulated using the appropriate schema, as FM or SSB or VSB demodulator, just to name some examples. The base band signal, from DC to 4 kHz, as the audio signal, will appear after the demodulator and problems, increasing or decreasing the SNR, the power density, the bit rate, the compression and the security level as the need. The capability to redo itself is required and innovations in the antenna are Fundamental. The beginning of the effort to create a SDR (platform that can transmit RF signals over software applicative) has started with the idea to improve a COTS amateur radio platform, which already a DSP with FM and SSB digital modulation.

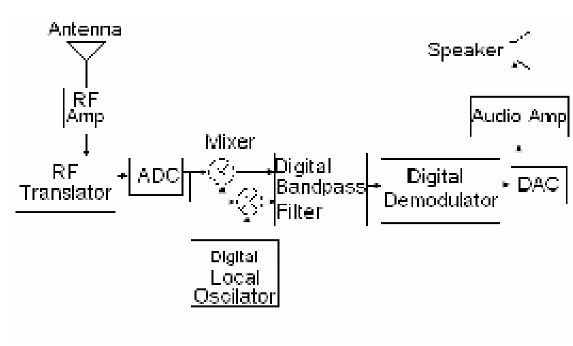

Fig. 2. Digital radio

The signal quality is kept because digital filters are easier to be created and the frequency response is better than in analog filters. At last, the combination of a heterodyne circuit with a digital software radio is more

be easily sampled with the commercial analog-digital converter. The suggested test bed will ratify the advantages and costs of these three sampling forms. Some comparisons with a commercial digital radio as shown in Fig.2 instead the proposed SDR platform will certainly be as referential datum.

B. Type II: Five port model

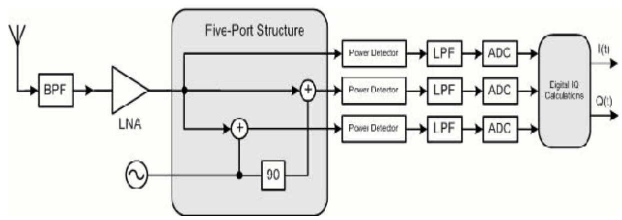

Fig. 3. Software defined radio direct conversion receiver using proposed five-port

To get a spars D-1 matrix, the first step is to eliminate D-1's elements. One of these relations is dependent of the others; so it can be removed and six ports will be reduce to five ports. By this way, the first column and the first raw of D-1 matrix will be eliminated. The second one is to set some elements as zero elements if possible. To get this objective, novel five-port architecture as shown in Fig.3 is proposed. This architecture uses a simple way to choose three independent outputs. This structure is very suitable for software defined radio applications. New structure reduced the number of ports to five ports, and it has a sparse D-1matrix due to zero setting of D-1's elements. Thus, there are less constant parameters to calibrate than the other structures. It is only need to determine five constant parameters. The parameter reduction in D-1 matrix makes easier the calibration procedure and causes to mitigate the errors of demodulation equations, which are appeared in calibration procedure. A new demodulator based on five-port architecture has been proposed and simulated. Its main advantages are the higher isolation between its two inputs and the less constant parameters in its demodulation equations than the previous multi-port structures. Simulation studies are conducted to evaluate the performance of the proposed demodulator. The digital modulation schemes are examined and BER of a specific digital modulation

(QPSK) is simulated. These results can be generalized to the MPSK and MQAM modulations. Furthermore, the simulation and measured results show that this architecture is comparable to the other multi-port structures and is more suitable for SDR terminals as result of its advantages such as broadband specifications, ultra low power consumption, less ADC in compare to six-port structure, high isolation between its two inputs and easy calibration procedure.

III. PROPOSED SYSTEM MODEL

A software defined radio is a transmitter and receiver system that uses digital signal processing (DSP) for coding, decoding, modulation, and demodulation. This allows much more power and flexibility when choosing and designing modulation and coding techniques. The Texas Instruments TMDSK6713 evaluation board with the TMS320C6713 DSP chip was selected to implement the radio. The system functions are shown in Fig. 4**.**

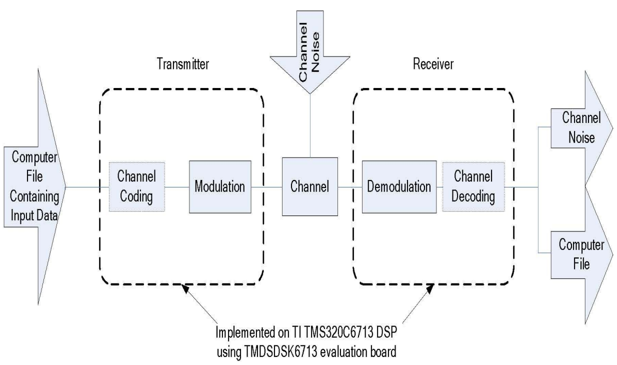

Figure 1 - I/O block diagram for transmitter and receiver radio systems

Fig. 4. I/O block diagram for transmitter and receiver radio systems

An overall block diagram for the software radio project is shown in Fig. 4. The inputs to the system are a digital data source (computer file) and channel noise. The output of the system is the recovered input data. The recovered data should be received exactly as transmitted. This can be displayed on an oscilloscope coming out of the DSP evaluation board and/or stored on a computer file for further verification and analysis.

The input from the digital data source will be sent into the transmitter. There it will have channel coding applied to provide protection from data corruption introduced by noise. This part will not be implemented in this project. After that, the encoded digital signal will then be modulated with an appropriate modulation technique and transmitted through the channel. An appropriate model and representation for the channel also needs determined. After this, the receiver demodulates the signal and applies appropriate channel decoding. From there the reconstructed digital signal will be available for further analysis.

Two development boards were used to construct the radio system. One board was designated for the transmitter and the other for the receiver. The system will be constructed and programmed entirely in

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off of the model designed, and the code was then downloaded to the board through TI Code Composer Studio for testing

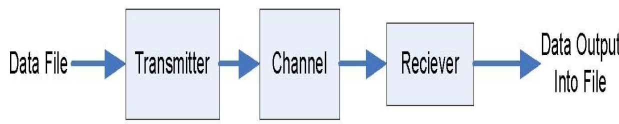

Fig. 5. System Breakdown of the Software Radio

The input to the system will be digital data in a computer file. This data will be modulated by the transmitter and sent to the channel. The channel will introduce interference to the signal in the forms of attenuation, phase delay, and noise. At the receiver side, the signal will be demodulated and reconstructed to produce the original transmitted message.

The transmitter shown in Fig. 6. will generate the signal that will be transmitted through the channel. The transmitter signal is constructed using demultiplexing, quadrature amplitude modulation (QAM), orthogonal frequency division multiplexing (OFDM), and up mixing.

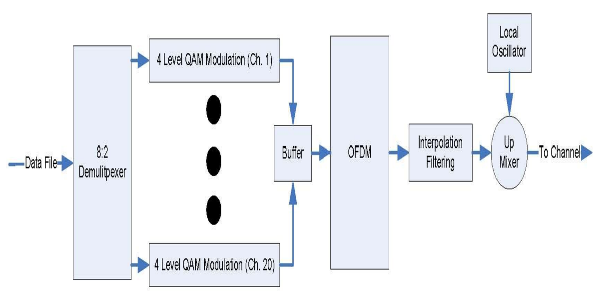

Fig. 6. Transmitter Subsystem Detailed Diagram

The demultiplexing block takes a byte of binary data and then breaks the byte into four 2-bit streams. These 2-bit streams are each fed into a QAM modulation channel. Once the QAM channels have modulated the input data, a buffer collects a group of 20 QAM symbols that represent 5 bytes of data. The group of symbols is then passed into the OFDM block. The OFDM system multiplexes the QAM signals together to produce the final modulated output. Interpolation increases the sampling rate and conditions the signal for transmission before it is modulated. This is implemented using a combination of up sampling and filtering. Mixing is done to meet the bandwidth requirements of the channel. The up mixer increases the frequency of the OFDM signal by multiplying it by a greater carrier frequency. The OFDM signal is imbedded in the carrier signal that the local oscillator produces. The output of the mixer is contained in bandwidth of the channel.

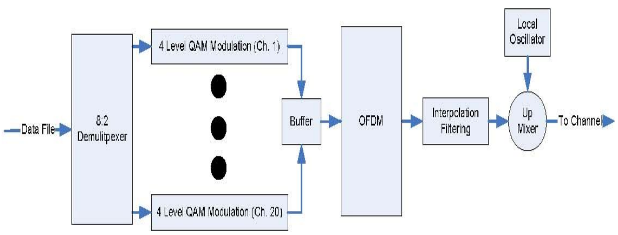

Fig.7. Channel Block

The channel block implements a model of an actual transmission channel. Different parts of the channel model different channel effects on the transmitter output. These are channel gain, multi-path interference, and noise.

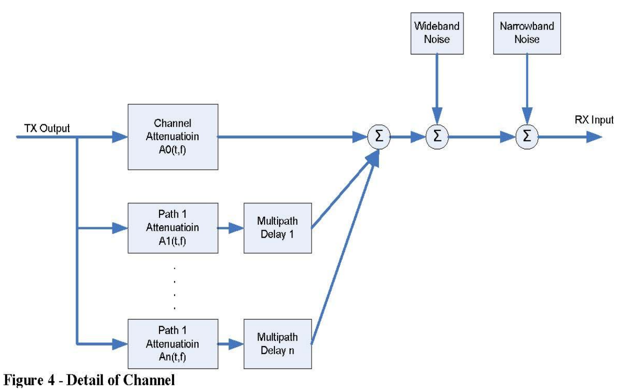

Fig. 8. Details of a channel

The channel attenuation models the attenuation or the gain effect that the channel has on the transmitted signal. This gain can vary with time and frequency. The multi-path interference models reflections of the transmitted signal. These reflections arrive at the receiver at different times. Each one of these paths has its own attenuation that can vary with time and frequency. Noise is also introduced into the signal. The noise is either specific to a limited frequency range (narrow band noise) or can affect the whole spectrum of the transmitted signal. The receiver subsystem shown in Fig.9 recovers the sent message. The receiver extracts the carrier, symbol, and frame timing from the signal. It uses this information to extract the message from the phase, frequency.

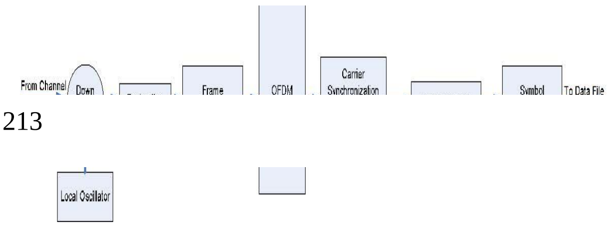

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Fig. 9. Receiver Subsystem Detail

The frame synchronization synchronizes the data frames. This aligns the start time of the message so the digital data can be interpreted correctly. This allows the compute file or message to be translated back into its original form. The demodulation system consists of two parts: OFDM demodulation and QAM demodulation. The OFDM demodulation demodulates the signal into its constituent QAM sub signals. The QAM demodulation decodes the QAM carriers back into the bytes that were originally transmitted. The carrier synchronization subsystem corrects for frequency differences between the transmitter and the receiver. It also corrects for the phase delay introduced by the channel. The symbol synchronization determines the time to sample the pulses coming from the QAM modulation. This allows the most accurate information to be extracted from the pulse stream. The output is the digital data that was originally sent into the system. What follows are the theory, design and implementation of each of the radio subsystems detailed earlier. Quadrature amplitude modulation is a modulation scheme that creates a modulation signal from a binary bit stream. The binary data is broken up into bit sets. Each bit set is represented on a constellation. The position of the point on the constellation representing the bit set is mapped to in-phase and quadrature components using the complex envelope. According to Couch [1], the complex envelope can be expressed as:

$$g(t) = x(t) + jy(t) \qquad (1)$$

In Equation 1, x (t) represents the in-phase and y(t) represents the quadrature component. Since the QAM in the software was at base band frequencies, mixing of the in-phase and quadrature parts of the QAM symbol was not needed. However, for transmission of a QAM symbol it must be mixed to higher frequencies for transmission, and can be represented as:

$$s(t) = x(t)\cos(\omega_c t) - y(t)\sin(\omega_c t) \qquad (2)$$

The data comes into the system one byte at a time. The byte of data is broken up into four pairs using the "extract bits" block. Once the bits are extracted, they are passed to a lookup table. The lookup table contains the QAM symbols representing each bit pair on the QAM constellation. The output of the QAM encoder blocks are complex numbers. The symbols are multiplexed together and converted into frame data. The buffer on the output of the convert to frame allows for multiple bytes to be placed into the OFDM spectrum.

IV. RESULT & DISCUSSION

A. Simulink

After completing the simulink model for software radio, it is tested by transmitting a test image through the transmitter model and received successfully. The test image transmitted through the model is given as in Fig. 10.

Fig. 10. Test Image (Transmitted image)

The transmitted image is viewed before transmission as in Fig. 11.

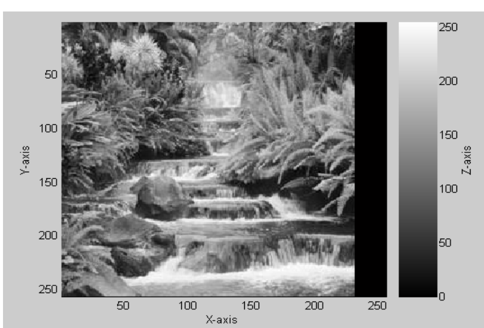

Fig. 11. Test image before transmission (Image in Viewer)

After transmitting through the model, the image is received successfully and is viewed at the out viewer which is shown in Fig. 12.

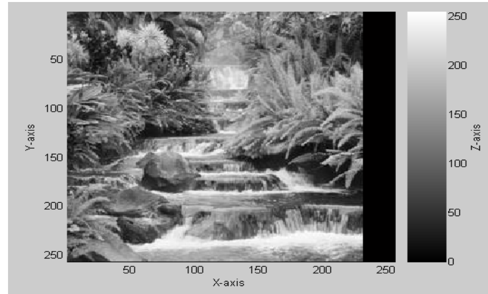

Fig. 12. Image out Viewer

Fig. 13. shown below defines that the picture is viewed while the image transmission occurs.

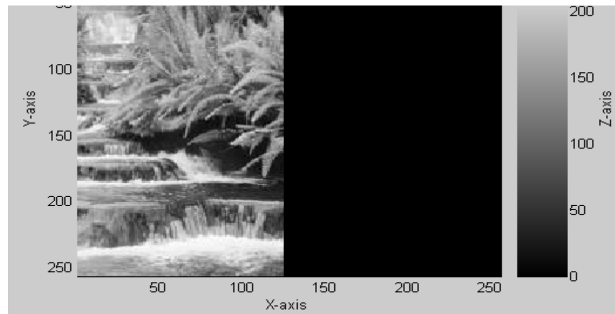

Fig. 13. Test image while transmitting

Fig. 14. shows the picture that is viewed at the time of reception.

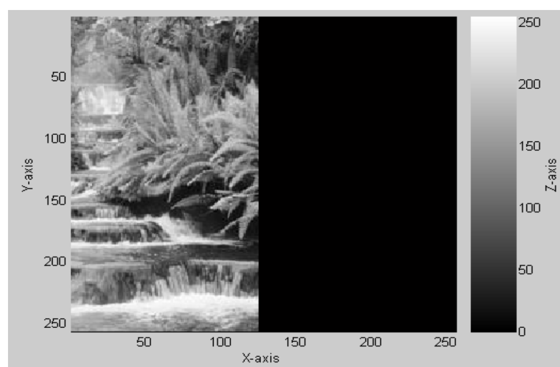

Fig. 14. Test image while reception

Fig. 15. describes unsynchronized OFDM Demodulation plot for the developed software model

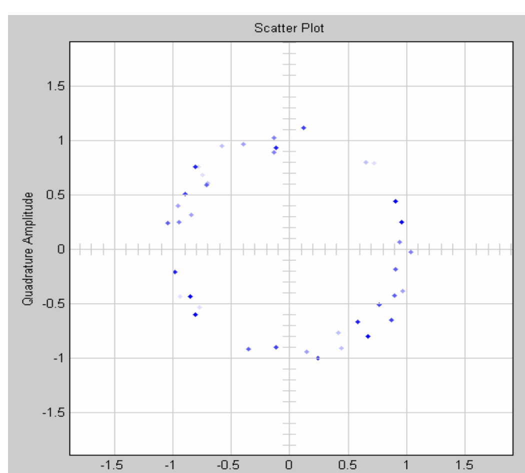

Fig. 15. Unsynchronised OFDM Demodulation Plot

Fig. 16. defines the OFDM Demodulation spectrum for the developed Software radio model.

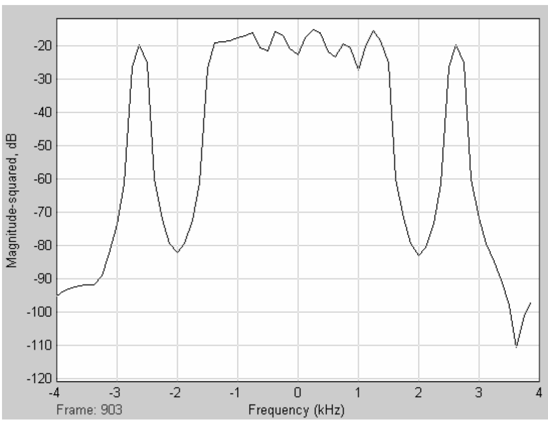

Fig. 16. OFDM Demodulation spectrum

Fig. 17. explains that the discrete time scatter plot scope obtained for the developed Software radio model.

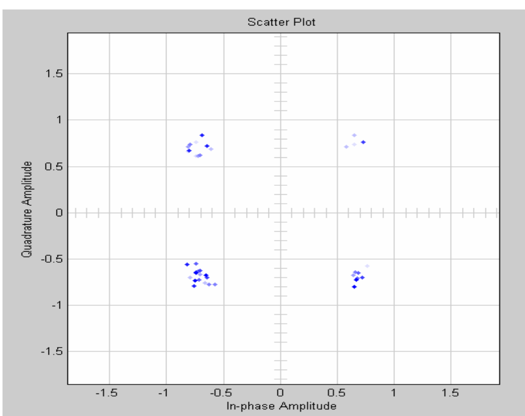

Fig. 17. Discrete time scatter plot scope

B. DSP Processor Board Results

the help of code composer studio, this has to be converted into DSP code which was tested by transmitting a text data through the processor board and is received successfully at the output. Fig.18

shows the main window application form for Rtdx interface.

8. Main window app

Fig.18. Main window application form for Rtdx Interface

Fig. 19. shows the transmission of text message through the processor board.

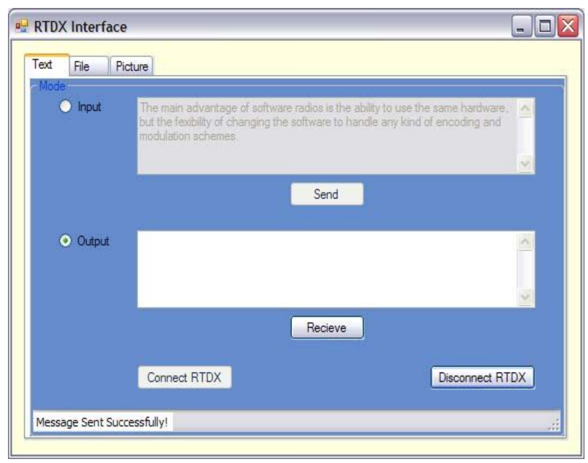

Fig.19. Transmission of text message through the processor board

Fig. 19. shows the reception of text message through the processor board with Rtdx.

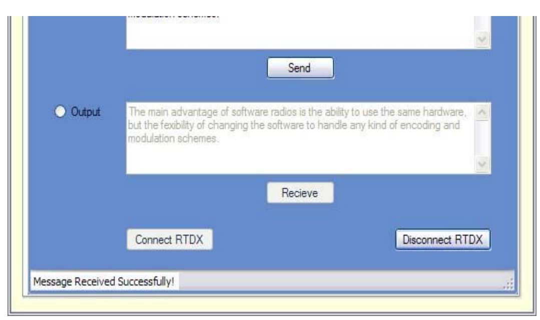

Fig. 20. Reception of Text message

V. CONCLUSION

Software Radio System was designed and implemented in simulink. The effects of potential channel disturbances were analyzed and compensated. The design was converted into C code and loaded to the DSP board. Though the code was too slow to run on the DSP board, the model is working efficiently for both jpeg image and for a text message.

Without the help of debugger, the C code can be converted to DSP code with the help of code compose studio, the utility provided by texas instruments. Comparing to other models, it is reliable and effective approaches to check the transmission of signal at each and every stage developed with the help of simulink and DSP processor board.

The model developed for verifying the transmission of a text jpeg image are,

- QAM encoder
- OFDM modulator
- QAM decoder
- OFDM demodulator
- OFDM spectrum
- Soft Radio Scatter Plot

Figuring out the simulink diagram optimization so that the radio speed could be increased to run on the board Implement channel coding on the radio to reduce the error rate due to additive white Gaussian noise. Looking into implementing RF hardware to simulate Radio's performance under real world conditions. Investigate Multilevel QAM and AGC in order to increase radio's data rate. The rest of the carriers of OFDM frame could also be utilized to increase this.

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