ABSTRACT: This paper proposed a STBC-OFDM system which is based on IEEE 802.16e OFDMA specification and supports the distributed subcarrier allocation of partial usage of sub channels (PUSC) for downlink (DL) transmission. The major parameters are summarized in Table I. The fast Fourier transforms (FFT) size is 1048. The length of cyclic prefix (CP) is 128 sampling periods. The modulation schemes of quadrature phase shift keying (QPSK) and 16 quadrature amplitude modulation (16QAM) are supported for data subcarriers, while binary phase shift keying (BPSK) is adopted for pilot subcarriers and preamble symbols. A DL sub-frame is composed of one preamble symbol and 40 OFDM data symbols. The system design target is to support carry frequency offset (CFO) up to 14 ppm and is optimized to enable the vehicle speed up to 120 km/hr. The proposed baseband receiver applied in the system with two transmit antennas and one receive antenna aims to provide high performance in outdoor mobile environments.

KEYWORDS: Space time block code – orthogonal frequency division multiplexing (STBC-OFDM) system, synchronizer, fast Fourier transforms (FFT), Modulation, Base band receiver.

I. INTRODUCTION

Most first generations systems were introduced in the mid 1980’s, and can be characterized by the use of analog transmission techniques and the use of simple multiple access techniques such as Frequency Division Multiple Access (FDMA). First generation telecommunications systems such as Advanced Mobile Phone Service (AMPS) only provided voice communications. They also suffered from a low user capacity, and security problems due to the simple radio interface used. Second generation systems were introduced in the early 1990’s, and all use digital technology. This provided an increase in the user capacity of around three times. This was achieved by compressing the voice waveforms before transmission.

Third generation systems are an extension on the complexity of second-generation systems and are expected to be introduced after the year 2000. The system capacity is expected to be increased to over ten times original first generation systems. This is going to be achieved by using complex multiple access techniques such as Code Division Multiple Access (CDMA), or an extension of TDMA, and by improving flexibility of services available.

The telecommunications industry faces the problem of providing telephone services to rural areas, where the customer base is small, but the cost of installing a wired phone network is very high. One method of reducing the high infrastructure cost of a wired system is to use a fixed wireless radio network. The problem with this is that for rural and urban areas, large cell sizes are required to get sufficient coverage.

Fig.1 shows the evolution of current services and networks to the aim of combining them into a unified third generation network. Many currently separate systems and services
such as radio paging, cordless telephony, satellite phones and private radio systems for companies etc will be combined so that all these services will be provided by third generation telecommunications systems.

Initial proposals for OFDM were made in the 60s and the 70s. It has taken more than a quarter of a century for this technology to move from the research domain to the industry. The concept of OFDM is quite simple but the practicality of implementing it has many complexities. So, it is a fully software project. OFDM depends on Orthogonality principle. Orthogonality means, it allows the sub carriers, which are orthogonal to each other, meaning that cross talk between co-channels is eliminated and inter-carrier guard bands are not required. This greatly simplifies the design of both the transmitter and receiver, unlike conventional FDM; a separate filter for each sub channel is not required.

Orthogonal Frequency Division Multiplexing (OFDM) is a digital multi carrier modulation scheme, which uses a large number of closely spaced orthogonal sub-carriers.

A single stream of data is split into parallel streams each of which is coded and modulated on to a subcarrier, a term commonly used in OFDM systems.

In OFDM, subcarriers overlap. They are orthogonal because the peak of one subcarrier occurs when other subcarriers are at zero. This is achieved by realizing all the subcarriers together using Inverse Fast Fourier Transform (IFFT). The demodulator at the receiver parallel channels from an FFT block. Note that each subcarrier can still be modulated independently.

Since Orthogonality is important for OFDM systems, synchronization in frequency and time must be extremely good. Once Orthogonality is lost we experience inter-carrier interference (ICI). This is the interference from one subcarrier to another. There is another reason for ICI. Adding the guard time with no transmission causes problems for IFFT and FFT, which results in ICI. A delayed version of one subcarrier can interfere with another subcarrier in the next symbol period. This is avoided by extending the symbol into the guard period that precedes it. This is known as a cyclic prefix. It ensures that delayed symbols will have integer number of cycles within the FFT integration interval. This removes ICI so long as the delay spread is less than the guard period.

This paper investigates the OFDM scheme, and realizes a fully functional system in software and analysing how it is reducing the inter-symbol interference caused by the multipath fading channels and different effects and estimating, evaluating the performance of it.

II. OFDM GENERATION

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the Orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required, based on the input data, and modulation scheme used. Each carrier to
be produced is assigned some data to transmit. The required amplitude and phase of the carrier is then calculated based on the modulation scheme (typically differential BPSK, QPSK, or QAM).

The required spectrum is then converted back to its time domain signal using an Inverse Fourier Transform. In most applications, an Inverse Fast Fourier Transform (IFFT) is used. The IFFT performs the transformation very efficiently, and provides a simple way of ensuring the carrier signals produced are orthogonal.

The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components. The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal.

The IFFT performs the reverse process, transforming a spectrum (amplitude and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length, which is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin. The orthogonal carriers required for the OFDM signal can be easily generated by setting the amplitude and phase of each bin, then performing the IFFT. Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal.

Fig. 2 OFDM Block Diagram

Fig. 2 shows the setup for a basic OFDM transmitter and receiver. The signal generated is a base band, thus the signal is filtered, then stepped up in frequency before transmitting the signal. OFDM time domain waveforms are chosen such that mutual Orthogonality is ensured even though sub-carrier spectra may overlap. Typically QAM or Differential Quadrature Phase Shift Keying (DQPSK) modulation schemes are applied to the individual sub carriers. To prevent ISI, the individual blocks are separated by guard intervals wherein the blocks are periodically extended.

III. SPACE-TIME BLOCK CODING

Multiple-input multiple-output (MIMO) system is known to exploit the antenna diversity to develop the performances of wireless communication systems using multiple antenna elements at the transmitter and receiver ends. The main objective of MIMO technology is to improve bit error rate (BER) or the data rate of the communication by applying signal processing techniques at each side of the system. The capacity increases linearly with the number of antennas while using MIMO however it gradually saturates. MIMO can obtain both multiplexing gain and diversity gain and can help significantly increase the system capacity. The earliest studies considering MIMO channels were carried out by Foschini [3] and Telatar.
MIMO can be divided into two main classes, spatial multiplexing (SM) and STC.

In a wireless communication system the mobile transceiver has a limited power and also the device is so small in size that placing multiple antennas on it would lead to correlation at the antennas due to small separation between them. To avoid this, the better thing to do is to use multiple transmit antennas on the base station and the mobile will have only one. This scenario is known as Multiple Input Single Output (MISO) transmit-diversity. A system with two transmit and one receive antenna is a special case and is known as Alamouti STBC. The Alamouti scheme is well known since it provides full transmit diversity. For coherent detection it is assumed that perfect channel state information is available at the receiver. However, when there is high mobility and the channel conditions are fluctuating rapidly it may be difficult to obtain perfect or close to perfect estimates for the channel. To alleviate this problem another space-time block coding techniques known as DSTBC has been proposed in [4]. In this technique, two serial transmitted symbols are encoded into phase differences and the receiver recovers the transmitted information by comparing the phase of the current symbol with the previously received symbol.

Transmit Diversity: Transmit diversity (TD) is an important technique to achieve high data rate communications in wireless fading environments and has become widely applied only in the early 2000s. Transmit diversity techniques can be categorized into open loop and close loop techniques [5]. For open-loop systems the most popular transmit-diversity scheme (depicted in Figure 4) is the (2x1) Alamouti scheme where channel state information and the code used is known to the receiver.

Alamouti Code: Alamouti system is one of the first space time coding schemes developed for the MIMO systems which take advantage out of the added diversity of the space direction. Therefore we do not need extra bandwidth or much time. We can use this diversity to get a better bit error rate. At the transmitter side, a block of two symbols is taken from the source data and sent to the modulator. Afterwards, the Alamouti space-time encoder takes the two modulated symbols, in this case x1 and x2 and creates an encoding matrix X where the symbol x1 and x2 are planned to be transmitted over two transmit antennas in two consecutive transmit time slots.

The Alamouti encoding matrix is as follows:

$$X = \begin{bmatrix} x_1 & x_2 \\ -x_2^* & x_1^* \end{bmatrix}$$

A block diagram of the Alamouti ST encoder is shown in Figure 4.2.
deliver a diversity order of $2 \text{Nr}$ [6]. Also, since for space time codes the rate is defined as $R=k/p$ (where $k$ is the number of modulated symbols the encoder takes as input and $p$ is the number of transmit antennas) for the Alamouti STBC the rate equals 1.

**Alamouti STBC Decoding with One Receive Antenna**

A block diagram of Alamouti STBC decoder is illustrated in Fig. 5. At the receiver antenna, the signals $r_1$ and $r_2$ received over two consecutive symbol periods can be written as follows:

$$r_1 = r(t) = h_1x_1 + h_2x_2 + n_1$$
$$r_2 = r(t + T) = -h_1x_1^* + h_2x_2^* + n_2$$

In order to estimate the transmitted symbols (two in this case) the decoder needs to obtain the channel state information (in this work we assume we have perfect CSI) and also use a signal combiner as could be seen from Fig 5.

**Fig 5 Alamouti Space-Time decoder**

The channel estimates together with the outputs from the combiner are then passed on to the Maximum Likelihood decoder (ML) to obtain the estimates of the transmitted symbols. Considering that all the constellation points are equiprobable, the decoder will choose among all pairs of signals $(x_1, x_2)$ one that would minimize the distance metric shown below:

$$d^2(r_1, h_1\hat{x}_1 + h_2\hat{x}_2) + d^2(r_2, -h_1\hat{x}_2^* + h_2\hat{x}_1^*)$$

$$= |r_1 - h_1\hat{x}_1 - h_2\hat{x}_2|^2 + |r_2 + h_1\hat{x}_2^* - h_2\hat{x}_1^*|^2$$

By substituting the above equation in below equation, the maximum likelihood decoding can be written as:

$$\hat{(\tilde{x}_1, \tilde{x}_2)} = \arg \min_{(x_1,x_2)} (|h_1|^2 + |h_2|^2 - 1) (|\tilde{x}_1|^2 + |\tilde{x}_2|^2) + d^2(\tilde{x}_1, \tilde{x}_2) + d^2(\tilde{x}_2, \tilde{x}_1)$$

Where, $C$ is all probable modulated symbol pairs $(x_1,x_2)$. $x_1$ and $x_2$ are formed by combining the received signals $r_1$ and $r_2$ with channel state information known at the receiver. The combined signals are given by:

$$\tilde{x}_1 = h_1^*r_1 + h_2^*r_2^*$$
$$\tilde{x}_2 = h_2^*r_1 - h_1^*r_2^*$$

Substituting $r_1$ and $r_2$ from above equations, the combined signals can be written as:

$$\tilde{x}_1 = (|h_1|^2 + |h_2|^2)x_1 + h_1n_1 + h_2n_2$$
$$\tilde{x}_2 = (|h_1|^2 + |h_2|^2)x_2 + h_1n_2 + h_2n_1$$

$h_1$ and $h_2$ are a channel realization, the combined signals $x_i$ i =1,2, depends only on $h_i$, i =1,2. It is possible to split the maximum likelihood decoding rule into two independent decoding rules for $x_1$ and $x_2$ as shown below:

$$\check{x}_1 = \arg \min_{(x_1)} (|h_1|^2 + |h_2|^2 - 1) |\check{x}_1|^2 + d^2(\check{x}_1, \check{x}_1)$$
$$\check{x}_2 = \arg \min_{(x_2)} (|h_1|^2 + |h_2|^2 - 1) |\check{x}_2|^2 + d^2(\check{x}_2, \check{x}_2)$$

Since for M-PSK modulated symbols $(|h_1|+|h_2| - 1) |2|$, i = 1, 2 is constant for all signal points equation (4.7) can further be simplified as:

$$\check{x}_1 = \arg \min_{(x_1)} d^2(\check{x}_1, \check{x}_1)$$
$$\check{x}_2 = \arg \min_{(x_2)} d^2(\check{x}_2, \check{x}_2)$$
Simple and Robust Synchronization

Synchronization includes symbol timing, sample clock, and carrier frequency synchronization. The proposed synchronizer concentrates on the symbol boundary detection and the carrier frequency recovery loop as presented in the following sections.

Symbol Boundary Detection: An ISI free region of symbol timing detection is determined by the difference in length between the CP and the channel impulse response. Since the proposed system has two transmit antennas, the signals transmitted from different antennas may arrive at the receiver with different delays due to multipath effect. Therefore, the decided boundary must locate in the common ISI free region to prevent the respective ISI effects from other symbols. IEEE 802.16 standard provides three types of preamble subcarrier sets which can be expressed as \[ s(t) = \max_n \left( \sum_{m=0}^{L-1} r[n+m] \cdot (p(t)[m])^* \right) \]

where and denote the sample indices. When the index corresponds to the peak of the matching results, the value of will be found. According to the values of 1 and 2, the final symbol boundary is decided to be located in the common ISI free region.

However, the mismatch of oscillator frequency in a receiver and a transmitter causes frequency offset effects in the received signals and destroys the characteristic of the matching results. In order to overcome this problem, we propose a modified match filter. The filter coefficient sequence which is the known preamble sequence is compensated with the possible values of ICFO. An output peak will appear in matching with the preamble sequence compensated with the corresponding ICFO. Hence, this match filter has an additional advantage that the coarse ICFO can be detected simultaneously. The proposed design is defined to support the frequency offset to 14 ppm variation. The maximum CFO is equivalent to 35 kHz in 2.5 GHz carrier frequency. Therefore, there are seven possible ICFO values ranging to.

Carry Frequency Recovery: After symbol boundary has been successfully obtained, the CP position is known. The CP repeating characteristic can be used to estimate FCFO by correlating the CP with the corresponding received sample sequence. However, when the CFO value is in the middle of two integer values, the accuracy of ICFO detection by using the match filter method is substantially decreased. Because the ICFO effects caused by these two integer values are almost the same, there are two undistinguishable peaks in the matching results. The undistinguishable peaks caused by noise and another antenna interference may easily result in wrong ICFO detection. Therefore, aping-pong algorithm is...
proposed to improve the performance of ICFO detection. The ping-pong algorithm partitions each CFO region into the strong region and the weak region depending on the distance to each integer value. The accurately estimated FCFO value can be used to correct the ICFO detection. When the estimated FCFO value locates in the strong region, the matching results have strong reliability to determine ICFO by detecting the peak value. When the estimated FCFO value locates in the weak region, there are two possible peaks in the matching results. Thus, the ICFO value will be adjusted by the information of the FCFO value and these two peaks. For example, there are two peaks appearing in the ICFO values of 0 and 1, and the FCFO value is correctly estimated to be 0.4 located in the weak region. If the ICFO value is detected to be 1 directly depending on the output peak, the estimated CFO is 1.4. This result is unreasonable because another peak is 0 but not 2. By using this ping-pong algorithm, the ICFO value will be correctly adjusted to be 0. The error probability of ICFO detection is simulated at the vehicle speed of 60 km/hr with a 16 dB ratio of the received bit energy to the noise power spectral density. The error probability using the direct detection method rapidly increases to about 0.5 when FCFO approaches 0.5 (the weak region), whereas it is normally about 0. However, the proposed ping-pong scheme is effective in maintaining ICFO error probability at about in the weak region.

**Accurate Two-Stage Channel Estimation**

Two major categories of pilot-aided channel estimation methods are interpolation-based channel estimation methods [4] and DFT-based channel estimation methods [7]. Interpolation-based method estimates channel frequency response (CFR) by interpolating the received pilot subcarriers. This method is not suitable for outdoor fast fading channels. This is because the channel coherent bandwidth becomes small, and using the interpolation-based method with limited pilot information becomes more difficult to recover channel variations. DFT-based method focuses on transform domain to characterize time-domain channel impulse response (CIR) and effectively improves the performance by suppressing time-domain noise. Many DFT-based methods derived from maximum likelihood (ML) scheme have been studied for OFDM systems with preambles [8]. In order to further improve the estimation performance, the DF DFT-based channel estimation method is employed by using decided data subcarriers as pilot subcarriers to track channel variations for saving transmission bandwidth and providing sufficient tracking information.

![Proposed Two stage Channel Estimator](image)

*Fig. 6 Proposed Two stage Channel Estimator*

A two-stage channel estimation method is used to realize a successful STBC-OFDM system in outdoor mobile channels. The multipath fading channel is characterized by CIR consisting of a few significant paths. The delays of these paths usually vary slowly in time, but the path gains may vary relatively fast. Therefore, in the initialization stage, the significant paths are identified during the preamble symbol time. In the tracking stage, the path gain variations in the identified path positions will be tracked in the following data.
symbol transmission. Fig. 6 shows the architecture of the two-stage channel estimator.

1) Initialization Stage: The operation blocks in this stage are a preamble match, an IFFT, a straight multipath interference cancellation (SMPIC)-based decorrelator, and an FFT. The preamble match correlates the received signal with the frequency domain preamble symbol to get the preliminary CFRs of different antenna pairs. Then, the CIR can be obtained.

The proposed match filter applied with the ICFO-compensated coefficients can reduce the CFO effect and provide the precise symbol boundary detection, and the ICFO value can be detected simultaneously.

- The proposed ping-pong algorithm using the estimated FCFO value can refine the ICFO value and improve the accuracy.
- The proposed two-stage channel estimation can highly improve the performance in outdoor mobile channels as compared with the interpolation-based methods that are frequently adopted in the baseband implementation.
- In the initialization stage, the proposed SMPIC-based decorrelator uses a straightforward method to identify significant paths and cancel the multipath interference, which can highly reduce the implementation cost.
- In the tracking stage, the matrix inverse computation is efficiently avoided by employing the strongly diagonal property, which can highly save the computation complexity.

IV. ARCHITECTURE AND CIRCUIT DESIGN

The decision of signal word lengths affects the system performance and hardware complexity. The output SNR at the STBC decoder is used as a performance criterion to determine the appropriate word lengths of each building block.

Synchronizer

In the match filter, each tap performs complex multiplication of the coefficient and the received sample. If the coefficients are quantized into, the tap operation can be simplified to add or subtract the received sample. It avoids the multiplier usage and reduces the register requirement. However, the sign calculation is necessary due to subtraction, and more hardware effort is required for sign extension. Therefore, if the signed calculation is avoided, the hardware and power will be further saved. In the proposed method, both the real and imaginary parts of the received sample are quantized into . The quantization loss is less than 0.8 dB in SNR, and it can be compensated by increasing the matching length , which costs only a little hardware overhead. When the total numbers of 0’s and 1’s stored in tap registers, denoted as and , are known, the number of -1 can also be obtained at the same time. Therefore, the matching result can be calculated as

Matching Result = \( N_1 - (L - N_0 - N_1) = 2N_1 - L + N_0 \).

According to the simulation results at the vehicle speed of 120 km/hr, the matching length of 300 is chosen in this case so that the symbol miss error probability approaches the lowest boundary. A monitor circuit is used to count after each data updating. The tap results are generated by the logical circuits and are accumulated to obtain by a carry save adder (CSA) tree. A logical circuit can be simplified to use only NAND gate and NOR gates. In the proposed symbol boundary detection, the coefficient sequence is pre-shifted with the possible values of ICFO, so the ICFO can be detected simultaneously. The coefficient sequence is stored in ROM. In order to accumulate the matching results twice for ICFO detection, an additional circuit is necessary to store the previous matching results as shown in Fig. 6. If the control signal is positive, the registers will respectively store
the current matching results of the possible ICFO values. On the contrary, if a control signal is negative, the chain of the registers works as the shift-registers for accumulating the matching results. Finally, the accumulated results are passed to the comparator to select the two ICFO values that have maximum peaks in the accumulated results.

![Fig. 7 Proposed Downlink base band receiver.](image)

![Fig. 8 Proposed OFDM system with two transmitters and one receiver.](image)

**V. RESULTS AND CONCLUSIONS**

The proposed baseband receiver with two transmitting antennas and one receiver antenna is simulated on Xilinx 13.2i and then they were implemented FPGA system. The proposed receiver effectively improves the design complexity with acceptable hardware cost while keeping the high performance in multipath fading channels. The Simulation results for the proposed system are shown in the below figure

![Fig.9 Bit error rate for the signal to noise ratio](image)

![Fig.10 Bit error rate with respect to the mobility of WLAN](image)

The above graph is the matlab implemented result for the ofdm architecture by increasing the fading noise between the transmitter and reciever .the graphs determines the linear decrease in the error rate by increasing signal to noise ratio.
The above simulation is to find the conjugate of the input and encoded using almouti scheme. Almouti scheme is discussed in the chapter 3. The format of the input is paved in the 2 transmit antennas.

**Fig. 11** Bit error rate with respect to noise ratio for different antenna number

**Fig. 12** Top module Simulation of Transmitter

The above simulation shows the input of the transmitter and the output of the STBC OFDM transmitter. The number of transmit antennas are two.

**Fig. 13** Simulation of STBC Encoder

**Fig. 14** Simulation of IFFT module

To make the serial serial data parallel ifft is implemented each for each transmitter.

**Fig. 15** Simulation of Receiver and signal detection module

The above is to check the guard band interval and implements the FCFO
methodology to correlate the both carrier wave in transceiver.

The proposed downlink baseband receiver for mobile WMAN that is applied in the STBC-OFDM system with two transmitter antennas and one receiver antenna. A simple symbol boundary detector, a carrier frequency recovery loop modified by the ping-pong algorithm, and an accurate two-stage channel estimator are effectively implemented. Although the two-stage channel estimator requires higher hardware cost as compared with the interpolation-based channel estimators, it has significant performance improvement for successfully realizing the STBC-OFDM system in outdoor mobile environments.

From the simulation results, we have shown that the proposed receiver improves about 8.5 dB of the normalized MSE for 16QAM modulation as compared with that adopting the 2-D interpolation methods in multipath fading channels. Moreover, under the vehicle speed of 120 km/hr, the convolutional coded BER for 16QAM modulation can achieve less than 10^-6 with coded rate of 1/2.

REFERENCES