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Analysis of LTE based Uplink Baseband Receiver in MIMO System

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***Abstract*— The present research ideas are moved towards the benefits of MIMO systems to next generation to achieving high data rates in future cellular standards after 3G (3rd Generation). In order to improve the increase speed and capacity of date rate we need advance Long Term Evolution (LTE) standard. LTE is based on spectrally efficient FDM (SEFDM) systems are Non-orthogonal overlapped carriers to improve spectral efficiency for future communication systems and reduce the PAPR due to its inherent signal structure. This paper presents the analysis of LTE based Uplink Baseband module in MIMO system.**

Keywords*— *LTE, MIMO, 3GPP, SEFDM

I. INTRODUCTION

The 3GPP long term evolution is a standard for wireless communication of high-speed data for mobile phones and data terminals. This LTE increasing the capacity and speed using a different radio interface together with core network improvements. The standard is developed by the 3GPP (3rd Generation Partnership Project).

In order to improve the high data rate transmission capability and high bandwidth efficiency in wireless communication system, the Orthogonal Frequency Division Multiplexing (OFDM) is necessary and it also improves the robustness to multi-path fading. It is a multi-carrier parallel data transmission technique, which partitions the spectrum into a number of sub-carriers modules, each one being modulated by a low-rate data stream. However, OFDM uses the spectrum more efficiently by spacing the channels much closer to each other [1-2]. It clubs all the sub-carriers orthogonal to each other. But this has many disadvantages includes the sensitivity to Carrier Frequency Offset (CFO) and large PAPR [3]-[4]. So researchers are more focusing on Single Carrier with Frequency Domain Equalization (SC-FDE) and SC-FDMA (Single Carrier Frequency Division Multiple Access systems) [5-6].

In order to accommodate for the ever growing demand for bandwidth, spectrally efficient FDM (SEFDM) systems emerged as multicarrier communication systems promoting higher spectral efficiency than the well-known orthogonal frequency division multiplexing (OFDM). The first systems to appear were Fast OFDM (FOFDM) [7] and m-ary amplitude shift keying OFDM (MASK) [8], both of which halve the spectrum utilization, but are constrained to one dimensional modulations such as BPSK and M-ary ASK.

All variants of SEFDM systems are basically multicarrier modulation schemes that multiplex non-orthogonal overlapped sub-carriers. In principle, non-orthogonal multicarrier systems achieve spectral savings by either reducing the transmission time and/or spacing between the subcarriers in frequency. Thus, communicating information at a faster than Nyquist rate. In theory, such spectral utilization improvement is supported by the Mazo limit established in [9] stating that signaling at rates beyond the Nyquist can be achieved without performance degradation.

The paper is organized as follows. Section 2 presents the overview of LTE System in words. Section 3 gives a review of the related previous work. A conclusion is given in Section 4.

II. OVERVIEW OF THE LTE SYSTEM

The LTE uplink receiver that integrates several advanced algorithms and features as shown in fig 1. It mainly consists of LDPC Encoder and decoder, Modulation techniques, sub-carrier mapping-demapping, DFT-IDFT models and sphere detector.

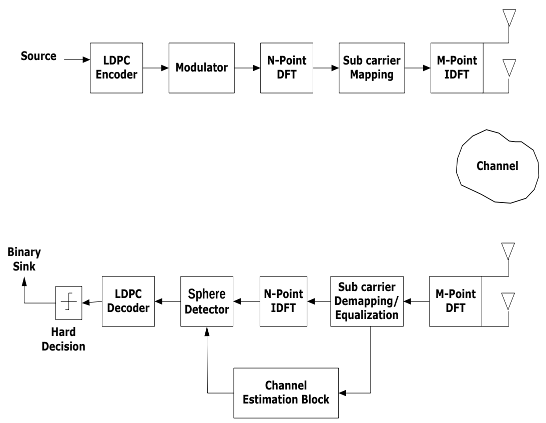

Fig 1. Model of MIMO system in LTE uplink Baseband receiver systems.

The source provides input binary data to the encoder. The LDPC Encoder contains their parity check matrix and the generator matrix is generally unknown and first encoded with an error correcting code [10-11]. It produces the encoded data. The 16-Quadrature amplitude modulation is a combination of amplitude and phase shift keying, so that a maximum contrast between each signal unit (bit, dibit, tribit, and so on is

achieved and produces modulated data.N-Point DFT convert the sampled function from its original domain (often time or position along a line) to the frequency domain.
Subcarrier maps the data to I-DFT.The Inverse discrete Fourier Transform block converts frequency domain to time domain. The output of I-DFT goes to Channel means data add up with additive white Gaussian noise and it is corrupted data. This data goes to MIMO Receiver.
The received data sequence is first transformed into the frequency domain by DFT. The subcarriers are demapped into sequence data or use minimum mean-square error ((MMSE) equalizer is a type of equalizer operating in both time and frequency domains. This equalizer can achieve better bit error rate (BER) performance with much higher algorithmic complexity. Because of the simple architecture and relatively good performance. The frequency domain sequence data is transformed back to time domain by IDFT. Detector detects the complexity of the sequence data and reduces the BER. The decoder decodes the correct data using correction techniques.

III. REVIEW OF LTE BASED UPLINK BASEBAND RECEIVER

The receiver module mainly contains DFT-IDFT Modules, Demapping, equalizer, detector and Decoder.

A. DFT and IDFT:

The SC-FDMA system [3-5] can handle up to Q orthogonal source signals with each source occupying a different set of M orthogonal sub-carriers. In the notation of Fig. 1, xm (m = 0, 1. . . M − 1) represents modulated source symbols and Xk (k = 0, 1. . . M − 1) represents M samples of the DFT of xm. Yl (l = 0, 1. . . N − 1) represents the frequency-domain samples after sub-carriers mapping and yn (n = 0, 1. . . N − 1) represents the transmitted time-domain channel symbols obtained from the IFFT of Yl. The sub-carriers mapping block in Fig. 2 assigns frequency-domain modulation symbols to sub-carriers.

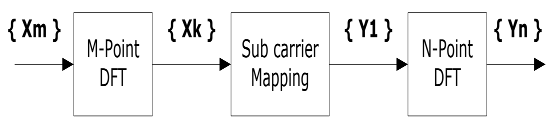

Fig 2: Generation of SC-FDMA transmit symbols.

There are N sub-carriers among which M < N sub-carriers are occupied by the input data. M, N: number of data symbols.
The signal after DFT can be expressed as follows:

$$X_k = \sum_{m=0}^{M-1} x_m e^{-\frac{j2\pi}{M}mk} \quad \text{----------- (1)}$$

Where M is the DFT length. After IDFT, the signal can be expressed as follows:

$$y_n = \frac{1}{N}\sum_{l=0}^{N-1} x_l e^{\frac{j2\pi}{N}nl} \quad \text{------------- (2)}$$

Where xl represents the signal after sub-carriers mapping, and N is the IDFT length. N >M. The IDFT in Fig. 1 and Fig. 2 creates a time-domain representation, yn, of the N sub-carriers symbols.
OFDM signal is efficiently implemented using IDFT. For an N carrier OFDM system, the input symbols are fed into a ρN point IDFT, where ρ is the number of samples per carrier, and then the outputs of the IDFT are fed into a digital to analogue converter to generate the continuous time domain signal. In [15] The IDFT implementation for SEFDM proportional transmitter. Zeros are inserted after the input symbols to suppress unwanted frequency and they designed Multiple IDFTs reduced Complexity implementation for SEFDM transmitter and receiver. This design eliminates the need for a bank of analogue modulators, allowing for an easy generation of SEFDM signals for a flexible range of frequency separation.
In [14] to SEFDM system, They map the algorithm to two variants of VLSI architecture, one with parallel IFFTs and one where they apply the concept of multi stream FFT to realize the multiple transforms at minimal circuit area overhead. A number of optimizations due to transforms on sparse input vectors are described to further reduce the number of arithmetic operations and FIFO sizes using a novel token flow approach and they are designed the IFFT and FFT using butterfly structure .
Implementation of radix-2^2 single-path delay feedback pipelined FFT/IFFT processor in paper [16]. This attractive architecture has the same multiplicative complexity as radix-4 algorithm, but retains the simple butterfly structure of radix-2 algorithm. The multipliers, Adder/sub tractor units, control unit, and their pipelining were implemented by efficient inferring the DSP48E Blocks in order to obtain a faster and low power design. The data and twiddle factor word length were chosen to achieve an acceptable signal-to-noise ratio and also to match the feature of DSP48E slices. The design can maintain the SNR since scaling and rounding are applied in all pipeline stages. The concept of Radix-2 serialized FFT (DIT) algorithm has been introduced to enhance the speed and area efficiency [17].

B. Subcarrier Demapping:

The received signal is converted to the frequency domain by the FFT (Fast Fourier Transform) block. Data sub carriers are identified and QPSK or QAM symbols are demodulated by the de-mapper block.
The set of values that each symbol can take is called the signaling alphabet, or constellation. Plotting the constellation in a two-dimensional plot can be done, with the x-axis denoting the real part bc[n] (corresponding to the I component) and the y-axis denoting the imaginary part bs[n] (corresponding to the Q component). Indeed, this is why linear modulation over passband channels is also termed two-dimensional modulation. Note that this provides a unified description of constellations that can be used over both

baseband and passband channels: for physical baseband channels, we simply constrain b[n] = bc[n] to be real-valued, setting bs[n] = 0.

Figure. Shows some common constellation mapping techniques. Demapping is exacting the original subcarrier by using mapping techniques. Pulse Amplitude Modulation (PAM) corresponds to using multiple amplitude levels along the I component (setting the Q component to zero). This is often used for signaling over physical baseband channels. Using PAM along both I and Q axes corresponds to Quadrature Amplitude Modulation (QAM). If the constellation points lie on a circle, they only affect the phase of the carrier: such signaling schemes are termed Phase Shift Keying (PSK). When naming a modulation scheme, we usually indicate the number of points in the constellations. BPSK and QPSK are special: BPSK (or 2PSK) can also be classified as 2PAM, while QPSK (or 4PSK) can also be classified as 4PAM.

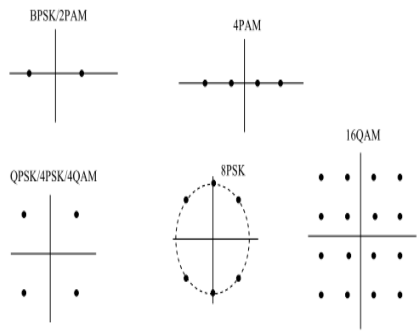

Fig: Some commonly used constellations demapping. Note that 2PAM and 4PAM can be used over Both baseband and passband channels, while the two-dimensional constellations QPSK, 8PSK and 16QAM are for use over passband channel.

Each symbol in a constellation of size M can be uniquely mapped to log2 M bits. For a symbol rate of 1/T symbols per unit time, the bit rate is thereforelog2 M/Tbits per unit time. Since the transmitted bits often contain redundancy due to a channel code employed for error correction or detection, the information rate is typically smaller than the bit rate. The choice of constellationfor a particular application depends on considerations such as power-bandwidth tradeoffs and implementation complexity.

Most of the papers [2, 6-8, 12], the demapping taken into account are BPSK, 4-QAM and 16-QAM for their work. Depending on the demodulation the amount of N bits 16 Implementation of a receiver System: 1 bit for BPSK, 2 bits for the QPSK and 4 bits for 16-QAM.

In communication systems[13], the iterative demapping and decoding techniques over quasi-static fading channels with significant diversity mappings traditionally optimized for iterative receivers (e.g., anti-Gray or Boronka mappings) outperform mappings more appropriate for non-iterative receivers (e.g., Gray mappings), in systems with limited diversity Gray based mappings in fact perform better than anti-Gray or Boronka mapping based schemes. In [13], they are used 16-QAM mapping schemes: Gray (top), anti-Gray (middle) and Boronka (bottom) mappings.

C. Equalization:

We use equalization to eliminate the effect of ISI and it is present due to the intentional violation of the sub-carriers orthogonality. There are several techniques for equalization such as Zero Forcing (ZF) equalization, Minimum Mean-Square Error (MMSE) equalization, Decision Feedback Equalization (DFE), and turbo equalization [20]-[24].

Linear detectors, such as ZF and Minimum Mean Squared Error (MMSE) are simple to implement but lead to significant Bit Error Rate (BER) degradation [25]. The ZF equalizer perfectly eliminates the effect of the channel in the absence of noise, but when noise cannot be ignored, the ZF equalizer suffers from the noise enhancement phenomenon. On the other hand, the MMSE equalizer takes into account the SNR.

Decision Feedback Equalizer (DFE) is widely used in the high-speed packaging system as a part of receiver for recovering the signal from the distortion by the inter-symbol interference (ISI). Decision feedback equalization (DFE) gives better performance for frequency-selective radio channels than linear equalization. In conventional DFE equalizers, symbol-by-symbol data symbol decisions are made, filtered, and immediately fed back to remove their interference effect from subsequently detected symbols [26].

By combining single-carrier (SC) frequency domain equalization (FDE) and Turbo equalization, In [27], propose a low complexity adaptive Turbo space-frequency equalization (TSFE) structure for single-carrier (SC) MIMO block transmission. Performing equalization on each frequency in independently, the proposed blockwise low complexity TSFE achieves a tremendous complexity reduction over the symbol-wise TSFE. With the same bandwidth efficiency, the low complexity TSFE provides performance close to that of the symbol-wise TSFE, and better than that of TTDE (Turbo time-domain equalization). With a moderate code rate, it is shown both theoretically and numerically that SC TSFE significantly outperforms its MC TOFDM counterpart [28], at a comparable complexity. The performance gains of TSFE over TTDE and TOFDM increase with the increase of channel delay spread. The low complexity TSFE is also incorporated with an adaptive channel estimation scheme referred to as LMS-SCE, utilizing correlated frequency bins. The LMS-SCE based TSFE provides a performance close to the perfect CSI case, at a high convergence speed.

D. Channel estimation Techniques:

The channel estimation technique used to estimate the realization of the multi-path channel system in the receiver model. The multi-path channel effect has problem as the each subband is disturbed by a channel of different random phase and amplitude. The challenging problems in wideband receivers is the tracking the effect of the multi-path channel.

In many application of noise cancellation, the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly .There are different channel estimation techniques which can exploit the pilot tones frequencies to estimate the effect of the channel. Out of these estimation techniques are Least Square (LS), Least Mean-Square (LMS) and Minimum Mean-Square

(MMSE), In [18] both Minimum Mean-Square, and Least Square (LS) channel estimation techniques have been presented and implemented over a multipath faded channel.

The utilization of adaptive algorithms includes Least Mean Squares (LMS) and Normalized Least Mean Squares (NLMS) [36] adaptive filters have been used in a wide range of signal processing application because of its simplicity in computation and implementation.

To increase the convergence speed of the LMS algorithm, the NLMS and AP algorithms was proposed in [35]. The Recursive Least Squares (RLS)[37] algorithm has established itself as the "ultimate" adaptive filtering algorithm in the sense that it is the adaptive filter exhibiting the best convergence behavior.The convergence property of the FAP (Fast Affine Projection) and FEDS (Fast Euclidean Direction Search) algorithms is superior to that of the usual LMS, NLMS, and affine projection (AP) algorithms and comparable to that of the RLS algorithm [34].

E. Detector algorithms:

Implementation of the recently developed MIMO detector algorithms and these can be classified as ML-based detectors, V-BLAST type detectors, and MMSE-based detectors. These solutions, although interesting in concept, still fail to meet the stringent latency and throughput requirements of a practical system such as wireless system.

The linear MMSE MIMO detectors have significantly lower complexity than ML algorithms and their performance. The hardware friendly algorithms that avoid matrix inversion for linear MMSE MIMO detection. They assessed algorithm complexity in terms of number of operations and bit precisions in fixed point designs, while considering FPGA implementation where a fixed number of dedicated hardware multipliers are available and they suggested a dynamic scaling technique for modified Gram-Schmidt QR decomposition that increases the numerical stability of the fixed point design [30].

Maximum likelihood (ML) search is the optimum detection method, which minimizes the BER. This scheme assumes an exhaustive search over the set of all possible transmitted symbol vectors. However, the complexity of full ML search is too high. Even with modern silicon technology the full ML search is still not feasible, especially for the MIMO detection with multiple antennas and high modulation orders.

Sphere detection (SD) solves the complexity problem of ML detection with some acceptable performance loss. In [19], the authors proposed a computationally efficient SD and list sphere detection (LSD) to achieve near-capacity performance on a MIMO system.

Alternatively, In the V-BLAST architecture, a successive interference cancellation (SIC) and nulling algorithm is used to detect the transmitted symbols, such a decision feedback detection mechanism is combined with a channel dependent detection ordering process. An improved VBLAST scheme using soft-input, soft-output, and soft-feedback is presented in [29], where the authors propose to make the symbol decision by minimizing the power of the interference plus noise, given a priori probabilities of undetected layer symbols and a posteriori probabilities of past detected layer symbols.

The spherical algorithm allows the detector to evaluate only a small subset of the transmitted candidates, and still achieve near-ML performance with enough candidates to calculate accurate soft output information [19]. The soft output information is important for the performance of soft-input error correcting decoders which are typically applied after MIMO detection.

The main Sphere Decoding (SD) process is an iterative algorithm, traversing a tree until a complete path is found. On each iteration, a new estimate for the radius and the interval centre are provided with the aim of reducing the sphere size and converging to a solution, which provides the most accurate estimate of the transmitted symbols. The elements of the solution are determined by enumerating the possible lattice points of the sphere, which are determined by the current sphere radius [25].

F. Decoding techniques:

The efficient encoding techniques are Hamming, Reed Muller, Golay, BCH, convolutional and Reed Solomon codes and etc. The goal was to construct codes with good properties includes shortest distance and to find low-complexity decoding algorithms, which are able to perform optimal decoding for these families. The decoding algorithms of these codes were many standards and applications include particularly convolutional and Reed Solomon codes, and there exist efficient and fast hardware implementations of these decoders.

Maximum a posteriori probability (MAP) decoder is an integral part of the most exciting error correcting turbo decoders. High speed architecture for MAP decoder is an essential entity for the design of high throughput turbo decoder which is widely used in the recent wireless communication standards. The MAP decoder based on Bahl-Cocke-Jelinek-Raviv (BCJR) algorithm [32]. Throughput of the turbo decoder immensely depends on the performance and the design of high speed architecture of MAP decoder to meet the throughput specification of 3G and 4G standards.

In The LDPC Codes, for large blocksize, achieve a performance very close to the Shannon limit and with low-complexity iterative decoding by Believe Propagation. (BP) or the Sum-Product Algorithm (SPA) in [11].

The design of LDPC decoder architectures differs from the decoder design for other classes of codes, in particular turbo codes, in that it is intimately related to the structure of the code itself through its parity-check matrix. The iterative decoding process of both codes consists of two main steps:

- computing independent messages proportional to the *a posteriori* probability distributions of the code bits
- Communicating the messages.

The complexity incurred in both steps depends on how messages are communicated with respect to the process of computing the messages [31]. Optimizations at the code design level aim at decoupling the decoder architecture from the code properties by decomposing the parity-check matrix of the code into permutation matrices resulting in architecture-

aware LDPC codes. Reducing memory requirements and improving decoder throughput have been addressed algorithmically through a novel turbo decoding algorithm of LDPC codes [31]. Moreover, an efficient message update mechanism has in the form of a message processing unit that reduces the switching activity of the decoder.
Reed- Solomon (RS) coder has been widely used in the FEC systems and provides an excellent way for correcting both random and burst errors and is capable of efficient correction of errors in wireless applications to provide high performance solution for 802.16 based wireless communication system [33].

Alternatively, veterbi decoder has proven to be a very practical algorithm for forward error correction of convolutionally encoded messages. In veterbi decoder, different Methods for back trace unit to find the correct path and high frequency by using parallel operations of decoder units. By seeing all these complexity of these decoders increased with the increasing of the constraint length they design adaptive Viterbi decoder that uses survivor path storage with parameters for wireless communication [33].

IV. CONCLUSION

In MIMO system, the LTE receiver contains DFT-IDFT blocks, subcarrier mapping, channel estimation, detector techniques and decoding algorithms and which integrates several advanced algorithms and features. It supports higher modulation orders and NXN antennas. We have reviewed the different algorithms for each blocks which is feasible to achieve low complexity and high performance in the LTE Receiver System

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