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Designing of LTE Communication System Under Adaptive Modulation Schemes

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ABSTRACT:

The LTE standard uses three different modulation schemes to adapt to various channel conditions in order to improve achievable data rates. These modulation schemes are the QPSK, 16-QAM and 64-QAM. This paper presents an overview of a LTE digital communication system, designed in order to study the effects of the QPSK, 16-QAM and 64-QAM modulation schemes on the BER performance with an AWGN channel model. Different subsystems within the transmitter and receiver blocks are implemented. It is noted that the LTE system uses Turbo channel coding and bit level scrambling to offer reliable and secure services to the users. Depending on the assumed channel condition (clear, medium clear or noisy), the 64-QAM, 16-QAM or QPSK modulation scheme, on the transmitter side; as well as the corresponding demodulation scheme, on the receiver side; are automatically selected. Based on the recovered data bits, the obtained bit error rates are analyzed, compared and discussed.

Keywords: LTE; QPSK; 16-QAM; 64-QAM; AWGN, Turbo coding, Bit-level scrambling, BER.

I INTRODUCTION

Long Term Evolution has long been seen as the first advancement towards stronger, faster and more efficient 4G data networks. The technology under LTE can currently reach downlink peak rates of 100Mbps and uplink speeds of 50Mbit/s. The LTE technology is also a scalable bandwidth technology for carriers operating anywhere from 20 MHz town to 1.4 MHz. Long Term Evolution offers some excellent advantages over current 3G systems including higher throughput, plug and play compatibility, FDD (Frequency Division Duplexing) and TDD (Time Division Duplexing), low latency and lower operating expenditures. It also offers legacy modes to support devices operating on GPRS systems, while supporting seamless pass-through of technologies operating on other older cellular towers. The authors of the (Global system for mobile communications) and UMTS (Universal Mobile Telecommunications System). The technologies put forth by LTE will not only be implemented over time, they are designed to

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be scalable. This scalability means the company can slowly introduce LTE technologies over time, without disrupting current services. The focus of this project is then turned towards implementing a fully operational LTE digital communication by synchronizing and integrating its different subsystems. This study particularly evaluates the impact of both the channel conditions based adaptive modulation and the Turbo channel coding on the BER performance of the system. As opposed to other related works, this design explores the isolated effect of LTE changes in the modulation schemes on the BER of the system. It then after explores the combination effects of modulation schemes adaptation and Turbo channel coding on the reliability of the communication system evaluated by means of the obtained BER performance. A theoretical BER performance model for the AWGN channel model is first analysed before the simulated BER results are obtained from the simulation of the fully integrated LTE. The obtained simulation results for the three modulation schemes are analysed, discussed and compared to the theoretically expected results before being compared to each other.

II. LITERATURE SURVEY

The Third Generation Partnership Project (3GPP) is developing a long term evolution (LTE) for the WCDMA based air interface. Key requirements of LTE include packet data support with peak data rates up to 100 Mbps on the downlink and 50 Mbps on the uplink, a low latency of 10 ms layer-2 round trip delay, flexible bandwidths (up to 20 MHz), improved system capacity and coverage, and efficient VoIP support. Advanced technologies were selected to meet these requirements including OFDM for the downlink, DFT-SOFDM for the uplink, MIMO, and turbo coding. The selection of turbo coding was considered carefully during the study item phase of LTE to meet the stringent requirements. After comparing several high performance codes (e.g., turbo codes, LDPC codes, etc) on the basis of complexity, flexibility, and backward compatibility, it was decided to use the existing WCDMA turbo code with a new contention free (CF) turbo interleaver to allow efficient parallelization of the turbo decoder for high data rates. Following deliberations within the working group, the Almost Regular Permutation (ARP) and Quadratic Permutation Polynomial (QPP) interleavers emerged as the most promising solutions to the LTE requirements with QPP selected for LTE.

Channel coding is one of the most important aspects in digital communication systems, which can be considered as the main difference between analog and digital systems making error detection and correction possible. Error correction exists in two main forms: ARQ (Automatic Repeat Request) and FEC (Forward Error Correction). With ARQ the receiver requests retransmission of data packets, if errors are detected, using some error detection mechanism. In FEC some redundancy bits are added to the data bits, which is done either blockwise (so-called block coding) or convolutional, where the

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coded bit depends not only on the current data bit but also on the previous bits. In LTE both block codes and convolutional codes are used. There is also an enhanced coding technique used in LTE, called Turbo code, which has performances within a few tenth of a dB from the Shannons limit. Another feature of LTE, which is considered here, is link adaptation. Link adaptation is referred to a mechanism matching automatically transmission parameters to the channel. As an example for older systems of link adaptation the early versions of UMTS (Universal Mobile Telecommunication System) can be mentioned, where fast closed-loop power control used to support an almost constant data rate. In UMTS the UE (User Equipment) transmitter adjusts its output power in accordance with one or more Transmit Power Control (TPC) commands received in the downlink, in order to keep the received uplink Signal-to-Interference Ratio (SIR) at a given SIR target. In HSPA (High Speed Packet Service Access) and LTE the transmitted information data rate is adjusted dynamically to use the channel capacity efficiently.

Link adaptation and feedback computation

Link adaptation in LTE

In LTE, link adaptation is based on the Adaptive Modulation and Coding (AMC). AMC can adapt modulation scheme and code rate in the following way: • Modulation scheme: if the SINR (Signal-to-Interference plus Noise Ratio) is sufficiently high, higher-order modulation schemes with higher spectral efficiency (hence with higher bit rates) like 64QAM are used. In the case of poor SINR a lower-order modulation scheme like QPSK, which is more robust against transmission errors but has a lower spectral efficiency, is used. • Code rate: for a given modulation scheme, an appropriate code rate can be chosen depending on the channel quality. The better the channel quality, the higher the code rate is used and of course the higher the data rate. In LTE for data channels a Turbo encoder with a mother code rate of 1/3 is used. There is a Rate Matching (RM) module following the Turbo encoder, which makes it possible to get other code rates, if desired. Increasing and decreasing the code rate is done via puncturing and repetition, respectively. Both, puncturing and repetition are integrated in the Rate Matching module. In Fig. 1 the whole signal generation chain of the LTEs physical layer with Turbo coding and modulation modules can be seen, which are parts of the link adaptation system.

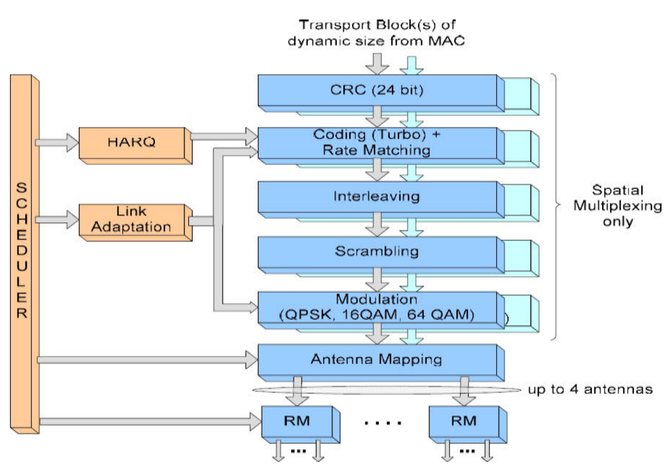

Figure 1: Signal generation chain in LTE

CQI feedback in LTE

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In LTE downlink, the quality of channel is measured in the UE and sent to the eNodeB in the form of so-called CQIs (Channel Quality Indicator). The quality of the measured signal depends not only on the channel, the noise and the interference level but also on the quality of the receiver, e.g. on the noise figure of the analog front end and performance of the digital signal processing modules. That means a receiver with better front end or more powerfull signal processing algorithms delivers a higher CQI. The signal quality measurments are done using reference symbols. Depending on the SNR (Signal-to-Noise Ratio) a combination of modulation scheme and code rate is selected to ensure that the BLER (Block Error Rate) is less than 0.1. This can be seen in Fig. 1

III EXISTING SYSTEM

All convolutional codes shall have a constraint length of 9. Convolutional encoding involves the modulo-2 addition of selected taps of a serially time-delayed data sequence. The length of the data sequence delay is equal to K-1, where K is the constraint length of the code.

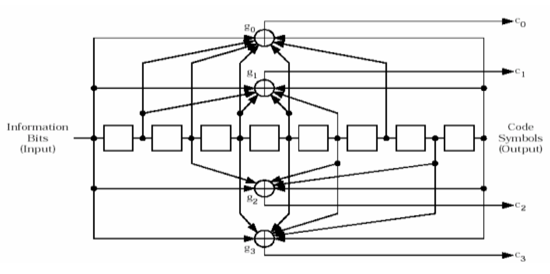

Fig 2: Convolutional Encoder

THE BASELINE WCDMA TURBO CODE

The turbo code (TC) in the WCDMA standard is a systematic code that consists of a parallel concatenation of two identical eight-state recursive convolutional codes interlinked by an interleaver. For an information block of size *K* bits, the turbo encoder generates a codeword of length 3*K* +12 bits, with a nominal code rate of 1/3 and 12 tail bits for trellis termination of the constituent encoders. The WCDMA TC supports all input block sizes between 40 and 5114 bits. Therefore over 5000 interleavers are defined based on an interleaving method that includes intra-row, inter-row permutations and pruning. Though the WCDMA TC interleaver is remarkable in that it ensures good performance at most of the 5000+ interleaver sizes, its performance is a concern at high code rates. Simulations indicate that the error floors of the WCDMA TC occur at Frame error rates (FERs), resulting in severe link-level performance loss. This performance inadequacy is due to the poor minimum distances of the interleaver at many block sizes; with puncturing, the distance properties are degraded even further. In contrast, recently developed interleavers such as DRP, ARP and QPP provide better performance than the WCDMA interleaver. This paper only discusses performance for the unpunctured rate-1/3 turbo code. The performance with puncturing for other code rates is considered as part of ratematching. For the interested reader, details on rate-matching can be found in the 3GPP RAN1 contributions. More importantly, the WCDMA turbo code

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interleaver does not allow an efficient decoder which satisfies LTE requirements. The WCDMA turbo code was designed to support data rates up to 2 Mbps while the peak data rates for LTE is 100+ Mbps. Although WCDMA decoder throughput can be increased beyond 2 Mbps by techniques such as increasing the clock rate or by radix-4 processing, replacing the WCDMA interleaver with a CF interleaver allows parallelized decoding with high throughput, low latency, and efficient hardware usage.

IV PROPOSED SYSTEM

The LTE system as illustrated in Fig. 1 comprises of:

- A transmitter block made of, from source to the channel, a formatter, a Mu-law compressor, CRC error detector, Turbo channel encoder, bit-level scrambler, NRZ baseband modulator and a selection based passband modulator subsystems.
- The AWGN defined by its noise variance parameter.
- A receiver block made of, from channel to destination, a passband demodulator, a bit-level descrambler, a Turbo channel decoder, a CRC error detector and a Mu-law expander subsystems.

V. METHODOLOGY

The simulation of the LTE communication system at its Physical Layer is crucial in order to assess and understand why and how the selection of a particular modulation scheme can affect its reliability in terms of its BER performance. One of the main distinguishing features of the LTE technology remains its ability to provide very high capacity and throughput services. In order to maintain such important features, the LTE system has to adapt its modulation scheme to the communication channel's conditions. This adaptation of the LTE modulation scheme impacts on the reliability of the system since it affects its BER performance.

Performing the QAM (Quadrature Amplitude Modulation) on the message signal

Quadrature amplitude modulation, QAM, when used for digital transmission for radio communications applications is able to carry higher data rates than ordinary amplitude modulated schemes and phase modulated schemes. As with phase shift keying, etc, the number of points at which the signal can rest, i.e. the number of points on the constellation is indicated in the modulation format description, e.g. 16QAM uses a 16 point constellation.

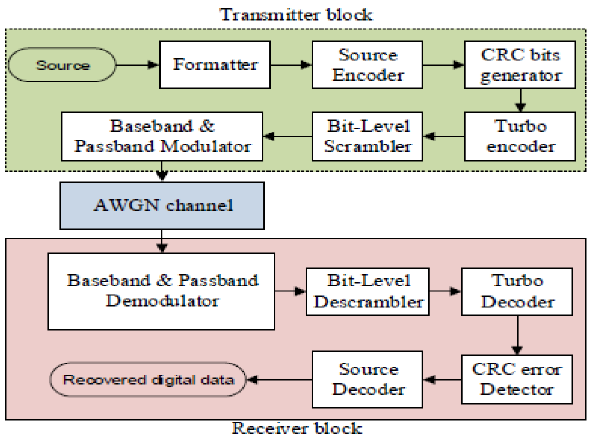

Fig. 3: LTE digital communication system block diagram.

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When using QAM, the constellation points are normally arranged in a square grid with equal vertical and horizontal spacing and as a result the most common forms of QAM use a constellation with the number of points equal to a power of 2 i.e. 4, 16, 64 . . . .

By using higher order modulation formats, i.e. more points on the constellation, it is possible to transmit more bits per symbol. However the points are closer together and they are therefore more susceptible to noise and data errors.

Normally a QAM constellation is square and therefore the most common forms of QAM 16QAM, 64QAM and 256QAM.

The advantage of moving to the higher order formats is that there are more points within the constellation and therefore it is possible to transmit more bits per symbol. The downside is that the constellation points are closer together and therefore the link is more susceptible to noise. As a result, higher order versions of QAM are only used when there is a sufficiently high signal to noise ratio.

To provide an example of how QAM operates, the constellation diagram below shows the values associated with the different states for a 16QAM signal. From this it can be seen that a continuous bit stream may be grouped into fours and represented as a sequence.

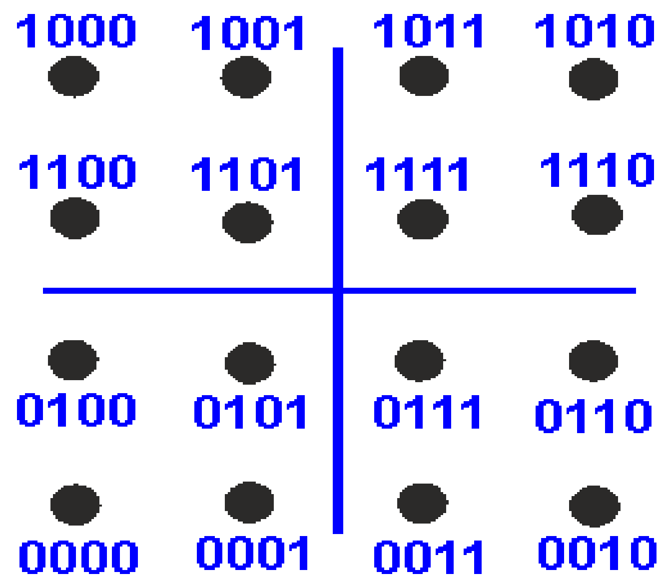

Fig 4: Bit sequence mapping for a 16QAM signal

Normally the lowest order QAM encountered is 16QAM. The reason for this being the lowest order normally encountered is that 2QAM is the same as binary phase-shift keying, BPSK, and 4QAM is the same as quadrature phase-shift keying, QPSK.

Additionally 8QAM is not widely used. This is because error-rate performance of 8QAM is almost the same as that of 16QAM - it is only about 0.5 dB better and the data rate is only three-quarters that of 16QAM. This arises from the rectangular, rather than square shape of the constellation.

VI. RESULTS

This section will present the simulation results and performance analysis of our proposed scheme. The presentation focuses on the recovery performance of our scheme in various situations.

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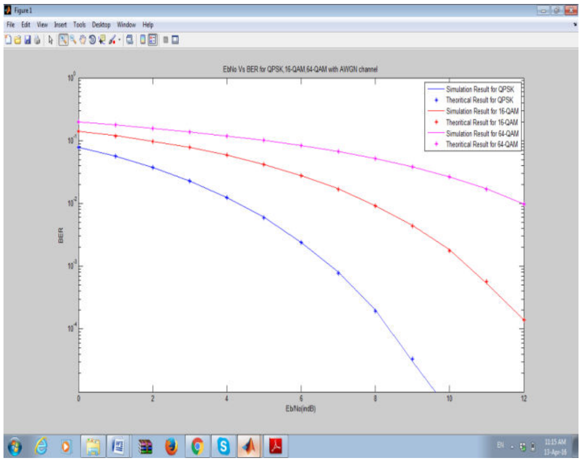

Fig 5: EbN0 vs. BER results for different modulation schemes like QPSK, 16-QAM and 64-QAM modulation schemes with AWGN channel.

From the above graph we can conclude that when the modulation order is increasing the BER performance will degrade.

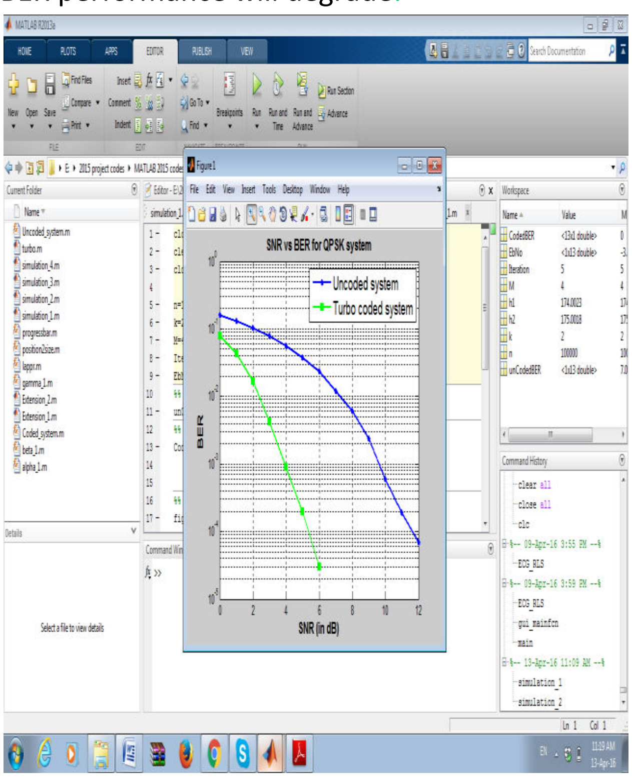

Fig 6: SNR vs. BER for QPSK system.

From the above graph we can conclude that always the coded system will perform better than uncoded system.

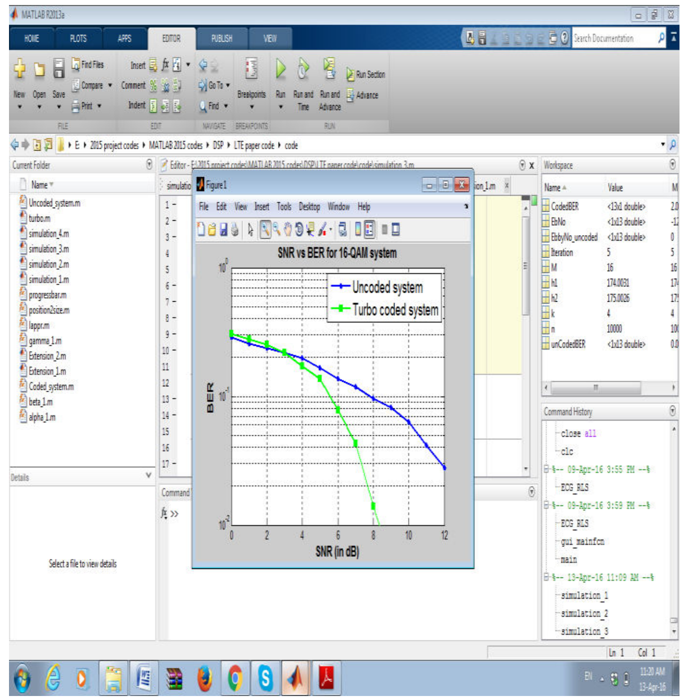

Fig 7: SNR vs. BER for 16-QAM system.

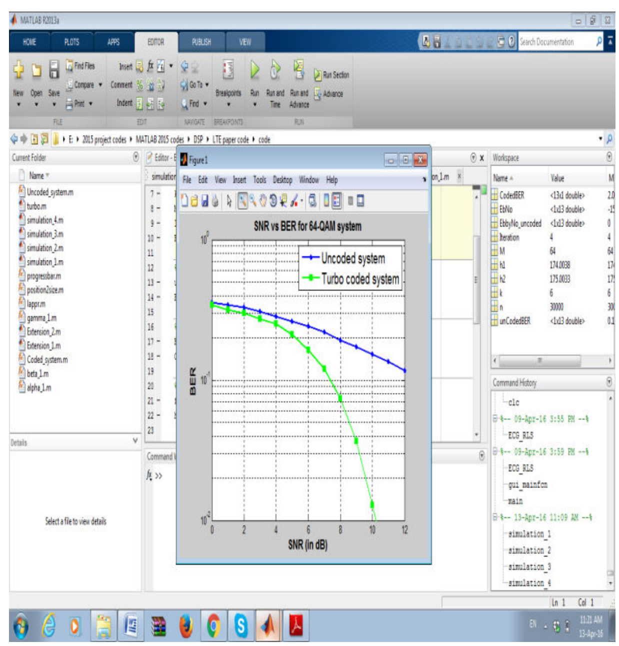

Fig 8: SNR vs. BER for 64-QAM system.

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From the above graph we can conclude that always the coded system will perform better than uncoded system

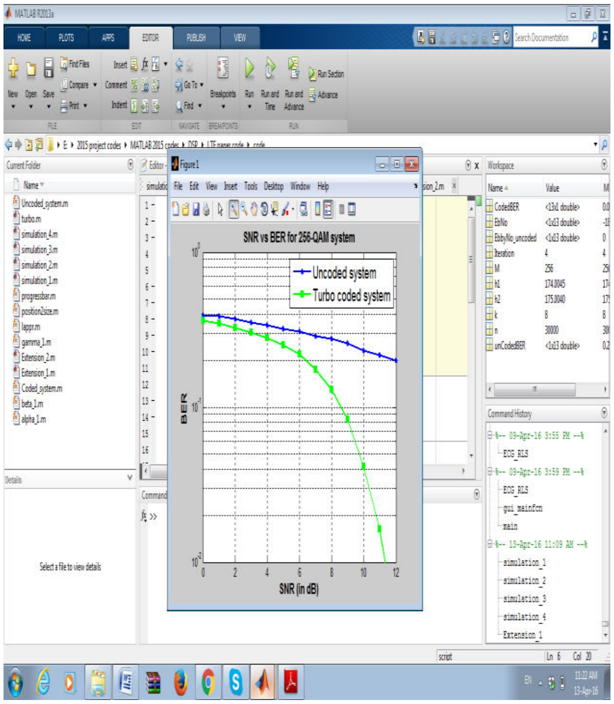

Fig 9: SNR vs. BER for 256-QAM system.

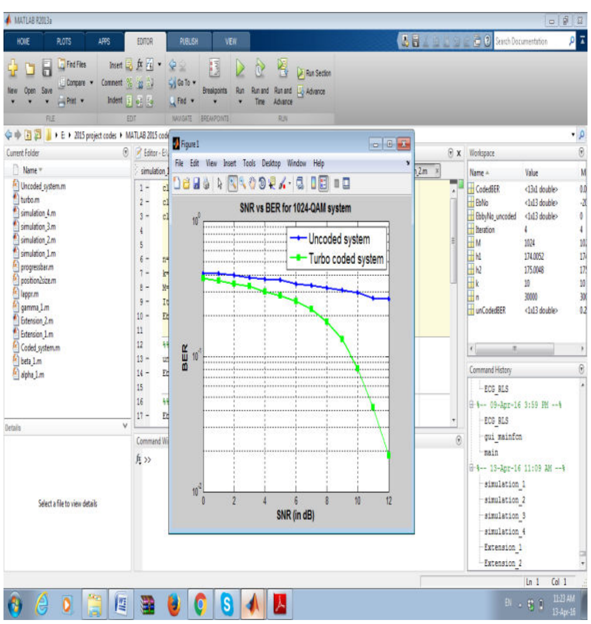

Fig 10: SNR vs. BER for 1024-QAM system.

VII. CONCLUSION AND FUTURE SCOPE

In this project, the design of a LTE digital communication system has been described. Different simulations of the designed LTE system have yield to different results. A comparison between the results obtained by simulating the LTE system without any channel coding subsystem and with the 1/3 Turbo channel coding has been established. The outcomes of the simulations have been analysed and it has been observed that the 1/3-Turbo channel coded LTE model performs much better in terms of BER than the non-coded model. It has also been observed that in both non-coded and 1/3 Turbo-coded scenarios, the denser the constellation modulation scheme (QPSK to 16-QAM to 64-QAM); the poorer its BER performance, meaning the poorer the reliability of the whole communication system. The benefit of our study to LTE industry and academia is to the prototyping tool and for the research and development laboratory.

In future, we can extend the results for even higher order modulation schemes and we will use more robust encoding schemes to improve the BER performance.

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