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High-Performance Mobile Technology LTE-A using the Stream Control Transmission Protocol: A Systematic Review and Hands-on Analysis

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Abstract: Long-Term Evolution-Advanced (LTE-A) is a new mobile radio standard of Long-Term Evolution (LTE) that uses orthogonal frequency division multiple access for the downlink and single carrier frequency division multiple access for the uplink. To relay nodes, LTE-A uses carrier aggregation aside from short delay time addition. This review attempts to shed some light on reviews and analyses performed on the Transmission Control Protocol (TCP) and Stream Control Transmission Protocol (SCTP) behavior over LTE/LTE-A to encourage researchers to contribute to this area. In particular, systematic models of various types of TCP are implemented and compared with the proposed SCTP protocol model. Given its high-rate data stream, LTE-A is considered the fourth generation of wireless mobile technology. A built-up visual diagram has highlighted LTE-A as a gene technology addendum enhanced version of LTE which is operated by frequency division duplex and time division duplex carriers. Nevertheless, multiple input and multiple output technologies have been presented and focused on the transport layer and protocols working underneath it. We conducted an in-depth review of the LTE-A features and how the TCP works on LTE/LTE-A. The strengths and weaknesses of two systematic models, namely, TCP and SCTP, are provided. Both systematic models are implemented to evaluate the performance of the protocols which highlighted the exchange process in congested environments. TCP seems to be in the middle of a crisis, especially with the development and growth of new wireless technologies, such as LTE-A. As such, this study proposed the use of the SCTP protocol because of its impressive architecture and unique functionalities.

Key words: LTE-A, SCTP, TCP, NS-2

INTRODUCTION

Long-Term Evolution-Advanced (LTE-A) which allows high data transfer rates, is an extension of the Long-Term Evolution (LTE) mobile standard. Improvements in LTE-A over LTE include higher bandwidth exceeding 1,000 Mb sec^{-1} and lower latencies.

LTE-A was officially nominated as a fourth generation (4G) system by the International Telecommunications Union (ITU) Telecommunication Standardization Sector. By the end of 2009, LTE-A was later confirmed by ITU. By March 2011, International Mobile Telecommunications Advanced (IMT-A) finalized the 3rd Generation Partnership Project (3GPP) as a major adoption of LTE. This approach was first implemented on October 2012 by the Russian mobile service provider YOTA (Dahlman *et al*., 2011; Rumney, 2013).

These working groups were formed by the radio communication assembly in 2007 and are also important for the development of LTE. International agreement has been achieved with respect to frequency distribution on the 7th World Trade Center in September 2009. The 3 GPP working group's proposed LTE Release 10 and later release (LTE-A) were submitted to the ITU. The previous LTE Release 8, including IMT-2000, have been replaced by ITU-R and officially applied in 2008 to satisfy the necessary requirements of the IMT-A catalog (Dahlman *et al*., 2011; Rumney, 2013). LTE-A was then presented at the Mobile World Congress 2011 in Barcelona. In June 2011, Ericsson LTE-A launched experiments to test the implemented commercial devices under the laboratory condition of 1.2 Gbps (De La Roche *et al*., 2012).

On October 2011, the proposal of the ITU-R Working Group WP5D was completed. The proposal fulfilled all the criteria necessary for approval and was officially adopted.

LTE-A as a new mobile technology can reach transfer rates of upto 1 Gbps which is significantly higher than

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those of the previous Universal Mobile Telecommunications System (UMTS, maximum transfer rate of 42 Mbps), the High Speed Packet Access [HSPA, also known as third generation (3G) technology] and other previous generations.

The LTE-A wireless standard, a further development of UMTS and HSPA, is a new mobile technology usually associated with the 4G technology. UMTS and HSPA belong to 3G technology. The Global System for Mobile Communications (GSM) with enhanced data rates for GSM Evolution is also known as IMT-Single Carrier and General Packet Radio System (GPRS); GSM is considered a second generation technology.

LTE is often referred to as a 3.9 G mobile network because it has not met certain specifications. Only the successor LTE-A is really classified as 4 G (De La Roche *et al*., 2012; Chen and Wang, 2010).

The activities in the physical layer of LTE-A have their own architecture category with regard to the data link layer and physical layer (Dahlman *et al*., 2011; Chen and Wang, 2010).

Thus, the LTE-A receiving and sending process is completed by network layers. Network layers have many classifications. The most well-known classifications are the Open System Interconnection (OSI) and the compact layers model Transmission Control Protocol and Internet Protocol (TCP/IP) (Kurose and Ross, 2013; Peterson and Davie, 2007).

Unequivocally, the network layers encounter problems, such as high bit rate exchanges, increasing number of users and increasing amount of end devices (Kurose and Ross, 2013).

Consequently, this review study focuses on a specific layer in the network layer, such as the transport layer or the harbor of shipping from the hard core network layers (physical, data link and network) to the soft core network layers (applications) (Kurose and Ross, 2013; Peterson and Davie, 2007).

The transport layer has an important part in the TCP/IP or IP process. Work inside the transport layer requires verification of protocols. LTE-A faces existing challenges in the transport layer, especially in its own protocol. Any protocol that works using LTE-A suffers from a huge amount of traffic. This study reviews the TCP in other places. The results of various TCP protocol models implemented were also reviewed.

The transport layer performs the transport protocol of the transport classes. Two communication application processes provide a transparent, seamless and secure end-to-end data transfer, regardless of the media used in layers 1 through 3. The following elements are the characteristics of the transport layer: transparency, accuracy, network independence, end-to-end transport service, cost optimization and transport addressing. Transport services can be categorized into two: Connection-oriented and connectionless. The datagram service is one example of the connectionless transport service (Peterson and Davie, 2007).

Conceptual framework: The conceptual framework is a template used to outline major points that have been covered across a research or to show a preferred way to an idea or thought. This study outlines the LTE technology because it is considered a parent of the LTE-A. As such, this study focuses on carriers, such as Frequency Division Duplex (FDD) and Time Division Duplex (TDD). This study also highlights the most important features of the LTE-A, such as Carrier Aggregation (CA) and Relay Node (RN) addition, to outline whole LTE-A offshoots, such as Multiple Input and Multiple Output (MIMO).

Likewise, this study illustrates the transport layer that belongs to the suite of network layers and covers well-known protocols inside this layer such as TCP and UDP.

Previous studies regarding TCP over LTE/LTE-A have analyzed and reviewed the two systematic models implemented to achieve accurate results through both models or across the results of the reviews (Fig. 1).

LTE-A: LTE-A is considered by many experts as an extension of LTE and the next generation of UMTS 3 G technology. LTE-A is not considered a surrogate of UTMS. Moreover, UMTS is based on GSM. LTE-A is rather an enhanced version of both UTMS and LTE and supplies a faster rate of downloading and uploading (Ghosh *et al*., 2010).

LTE-A was officially presented as a nominee of 4 G technology in late 2009 which was approved by the ITU. On March 2011, IMT-A categorized LTE-A as an enhancement of LTE and considered LTE-A as a 3 GPP (Dahlman *et al*., 2011; Ghosh *et al*., 2010).

Originally, LTE-A was considered to have evolved from LTE which in addition to WiMAX, is classified as 4G technology. LTE-A is different from LTE in a number of ways, including time delay and CA. Other differences are discussed in the succeeding paragraphs (De La Roche *et al*., 2012).

The properties of LTE-A (Rumney, 2013; Li *et al*., 2010) are as follows:

- Increased peak data rates of 1 Gbps downlink (DL) and 1 Gbps uplink (UL)
- High spectral efficiency of 16 bps Hz^{-1} in release 8 and 30 bps Hz^{-1} in release 10
- High number of concurrent users

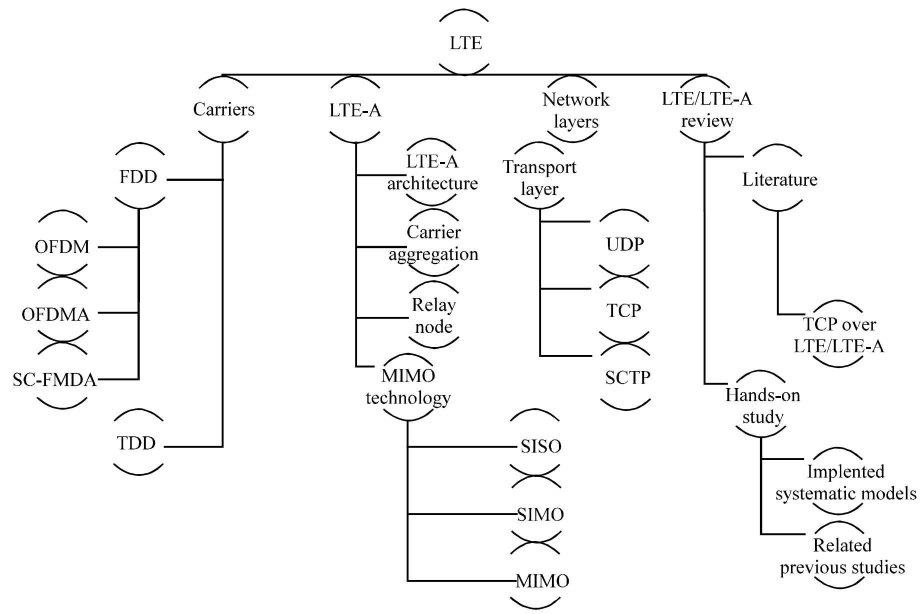


Fig. 1: Conceptual framework

- High data rates at the cell edge; for example, at least 2.4 bps/Hz/cell for UL 2×2 MIMO

The most important new features that have been introduced into LTE-A (Ratasuk *et al*., 2010) are as follows:

- CA
- Improved use of multi-antenna techniques (for DL 8×8 MIMO)
- Support of RN

FDD and TDD: LTE and LTE-A used the carrier frequency for the sending and receiving processes. FDD and TDD as access mode are two types of carriers that can be implemented inside LTE and LTE-A.

The FDD has a Downlink (DL) and an Uplink (UL) detected frequency band. By comparison, the TDD UL and DL share frequency over time (Rumney, 2013).

By using paired and unpaired spectra, the majority of systems operate by FDD. Meanwhile, TDD will be deployed by some providers later on this year (Fig. 2-3).

The frequency of FDD is patterned after UTMS frequencies. Some additional frequencies have been added since then (Dahlman *et al*., 2011).

Table 1 briefly shows the difference between FDD and TDD. An overlapping frequency or a subset frequency means that many of the existing power amplifiers and receivers are out of coverage.

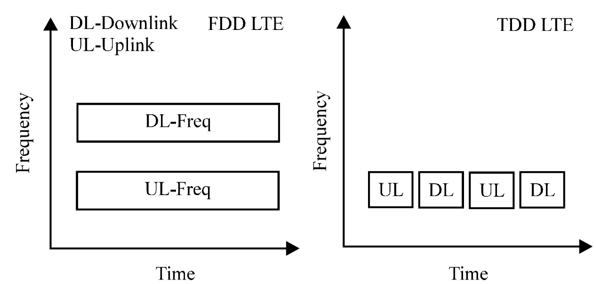


Fig. 2: FDD vs. TDD

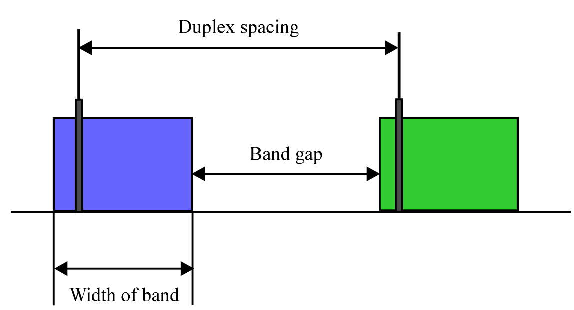
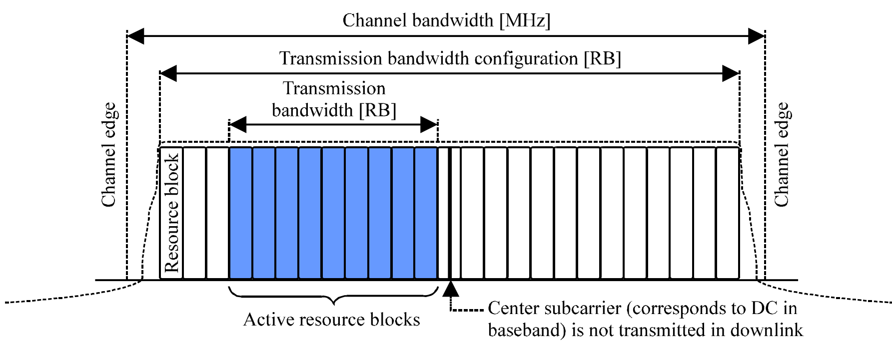


Fig. 3: FDD structure diagram

The narrow space between duplex spacing and space gap makes it difficult to design a filter to prudently transmit regrowth. However, the narrower duplex spacing and gap makes it difficult to design filters that prevent transmitter spectrum regrowth and adds complexity to the design of multiple antennas. Therefore, the ability to

Table 1: FDD and TDD approaches

FDD and TDD methods		
Characteristic	FDD	TDD
Spectrum usage	High including guard bands	Less
Complexity	High	Low but need accurate timing
Cost	Higher	Lower
Latency	Little or none	Depends on range, transmission-receive switching times
Range	Unlimited	Shorter, depends on time guard
DL/UL symmetry	Usually 50/50	Asymmetry as required
Dynamic bandwidth allocation	None	Can be implemented
MIMO and beamforming	More difficult	Easier



Channel bandwidth (MHz)	1.4	3	5	10	15	20
Transmission bandwidth configuration (MHz)	1.06	2.7	4.5	9	13.5	18
Transmission bandwidth configuration (RB)	6	15	25	50	75	100

Fig. 4: LTE-A TDD frequency bands

implement multiple bands in devices is an issue in customization (Rumney, 2013; Li *et al*., 2010).

Different bands consist of a DL band and an UL band. Both DL and UL bands have a width and a gap isolated between them. The download link bandwidth plus the gap is equal to duplex spacing.

Numerous regional overlaps between band definitions occur because of different reasons.

The UL uses a frequency different from that used by the DL, even though these frequencies are in the same band. For example, band 1 UL used 1920 and 1980 MHz and DL used 2110 and 2170 MHz; the width size is 60, the gap is 130 and the duplex is 190. The space of the duplex moved from 30-800 MHz. Similarly, the gap varied from 10-680 MHz, with the gap working to separate between UL and DL. Some bands are reserved for regional states, such as bands 16 and 17 which are specified for use in Europe.

The channel bandwidth is a flexible stretch from 1.4-20 MHz. Each channel bandwidth consisted of various numbers of Resource Block (RB). Each RB contains 12 subcarriers, with each carrier having a bandwidth of 180 kHz. As such, the channel bandwidth of 1.4 MHz has 6 RB, whereas 20 MHz has 100 RB (De La Roche *et al*., 2012).

The transmission architecture scheme includes two DL and UL processes. In the DL mode, LTE-A uses the Orthogonal Frequency Division Multiple Access (OFDMA). In the UL mode, LTE-A uses the Single Carrier Frequency Division Multiple Access (SC-FDMA).

Transmission Time Interval (TTI) is the time needed to transfer one block of data for LTE-A. The TTI is always approximately 1 msec. The lowest TTI is better because it is the shortest interval to transport a data block.

The TDD has no concept of duplex spacing or gap sensitivity for download link and upload link frequencies transmitted on the same band. As such, separating the transmission from receiver to sender switch and deploying the duplex filter that uses the frequency switch time domain become a challenge (De La Roche *et al*., 2012).

The transmission bandwidth is determined independently by the number of active subcarriers or resource bugs (Rumney, 2013; Chen and Wang, 2010). This transmission bandwidth gives the LTE the flexibility to have six different configurations that range from 1.4-20 MHz. The channel bandwidth is defined in megahertz which is phenomenally occupied by the channel transmission bandwidth defined by the RB (Tran *et al*., 2012) (Fig. 4).

OFDM: Orthogonal Frequency Division Multiplexing (OFDM) enables multiple transmission bandwidths. The OFDM is a multi-carrier modulation scheme wherein the radio signal is divided into many parts, such as sub-signals or subcarriers which transmit to each other. These subcarriers modulate various frequencies in a large number of compact subcarriers.

The modulation scheme of OFDM subcarriers traditionally takes the Quadrature Phase-Shift Keying (QPSK) format, with 16-64 Quadrature Amplitude Modulation (QAM).

The multi-carrier modulation scheme achieves transition efficiencies better than some traditional signal carrier schemes (Dahlman *et al*., 2011; De La Roche *et al*., 2012). However, this scheme is not immune to the permanent common channel. The single carrier modulation scheme has a higher symbol rate than multi-carrier modulation scheme with the same bandwidth.

The sine cardinal (Sinc) function of OFDM is augmented in LTE-A, with each carrier dependently represented by its own equation (Dahlman *et al*., 2011; De La Roche *et al*., 2012) (Fig. 5).

The Sinc function is often established in the signal process field and in LTE-A. Sinc(x) which is also referred to as the sampling function, is a function used in theory of Fourier transforms. We noted that the Sinc function is considered a vital operation of OFDM carriers (Dahlman *et al*., 2011):

$$\text{sinc}(x) = \begin{cases} 1 & \text{for } x = 0 \\ \dfrac{\sin x}{x} & \text{otherwise,} \end{cases}$$

(Rumney, 2013; De La Roche *et al*., 2012).

The Sinc of x is a sine function which is plotted in Fig. 6. In Fig. 6 the value of x ranged from -15 to 15.

OFDM operates as a number of orthogonal (non-interfering) narrow band systems composed of carrier spacing, phase noise, timing and frequency offsets working against the orthogonal system.

Each of these Sinc carriers has its own modulation symbol scheme that ends up as the modulation magnitude phase.

The number of subcarriers for each user depends on the data rate as data spread out over these subcarriers.

OFDMA: LTE uses OFDMA. OFDMA is more advanced than OFDM is. Subcarriers are allocated to different users over time. As such, OFDM user allocation is in the time domain only, whereas OFDMA user allocation is in the time and frequency domains (Lee *et al*., 2009) (Fig. 7).

In addition, OFDM uses all subcarriers of the symbol support data to determine the user. OFDMA uses subcarriers of the symbol support data that may be split among multiusers, thus covering the use of radio resources and, more effectively, the terms of use (Dahlman *et al*., 2011; Lee *et al*., 2009).

The dynamic allocation in the OFDMA stabilizes the use of the channel with regard to multiple low-rate users, preventing fading and interference issues in narrow bands.

Some benefits of OFDM include multiple subscriber scalable bandwidth channels, frequency-selective

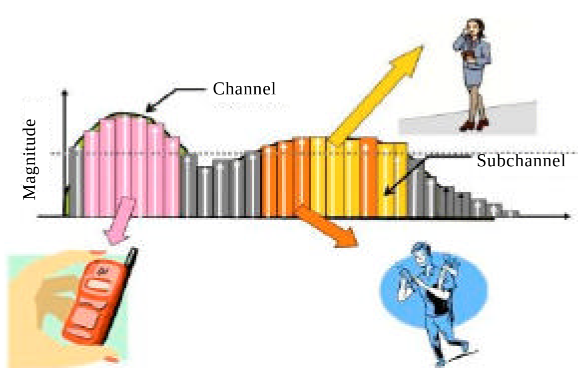


Fig. 5: OFDM

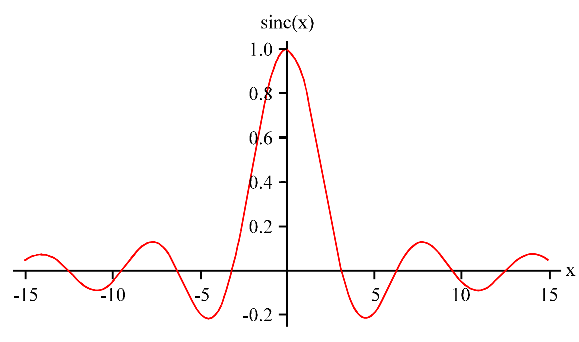


Fig. 6: Plotted sinc(x) function

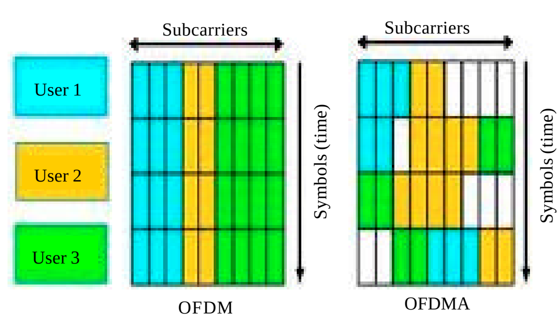


Fig. 7: OFDM vs. OFDMA

scheduling within channels, wide channel potential that supports high data rates and resistance to multi paths due to long symbols.

Nevertheless, OFDM has some limitations, such as sensitivity to frequency failures and phase noise due to close subcarrier spacing; sensitivity to Doppler switch which results in interference between users or subcarriers. A large amount of double sideband competition is needed to address interference (Yao *et al*., 2012).

The ideal OFDM reacts to a high Peak-to-Average power Ratio (PAR) in the time domain because the SC-FDMA is used for the UL.

SC-FDMA: SC-FDMA is one of the access modes used by LTE-A for the UL which has a hybrid transmission scheme. The DL uses multiple carriers of RB that cluster a relatively long symbol duration.

Each of the subcarriers is modulated by different data symbols and this modulation lasts for a relatively long duration (Rumney, 2013; Lee *et al*., 2009).

SC-FDMA has multiple subcarriers. However, all subcarriers in the UL are modulated by the same data. As shown in Fig. 8 (right side), the first group of green blocks belongs to the same data with physical RB. However, the cluster has a short time which means that the symbol will be short.

OFDMA modulates QPSK, with each M subcarrier encoded with one QPSK symbol. Note that M subcarriers are equal to four subcarriers. Therefore, M subcarriers can transmit M QPSK symbols in parallel. Figure 8 shows how OFDM modulates the QPSK symbol (Fig. 8).

With many subcarriers, the in-phase "I" and quadrate "Q" waveforms become Gaussian in the same context SC-OFDM modulates QPSK to transmit 1, 1 -1, -1 -1, 1 1, -1 and create a time domain representation of the IQ baseband sequence (Fig. 9).

We performed a discrete Fourier transform of length M and sample rate M/(symbol period) to create M Fast Fourier Transform (FFT) bins spaced 15 kHz apart.

Figure 10 shows M subscribers in the desired allocation within the system bandwidth. A time domain signal of the frequency that shifted from the original is also created by performing inverse FFT. Moreover, a cyclic prefix is inserted and transmitted between SC-FDMA symbols.

PAR is the same as the original QPSK data symbols which are different from those of the OFDMA. The OFDMA and SC-FDMA schemes have different behavior (Dahlman *et al*., 2011; Tran *et al*., 2012).

Cursory checking of the complementary cumulative distribution function shows that the OFDMA scheme has a longer probability curve than SC-FDMA does. SC-FDMA has lower PAR compared with OFDMA and extra headroom that lowers costs in the power amplifier and reduces battery drain.

LTE-A used the FDD or TDD to represent the physical layer. As such, the FDD uses the UL and DL transmitted separately (De La Roche *et al*., 2012; Kishiyama *et al*., 2013).

Each frame in FDD has a structure type that includes many sub-frames, with each sub-frame having a hash index. Each sub-frame has a specific time period which is the outcome of the time domain of one radio frame.

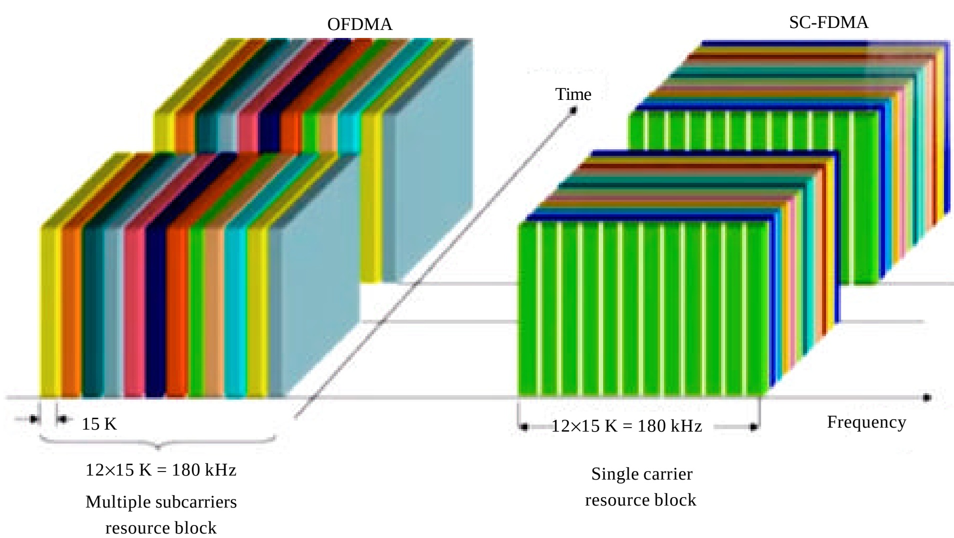


Fig. 8: OFDMA and SC-FDMA

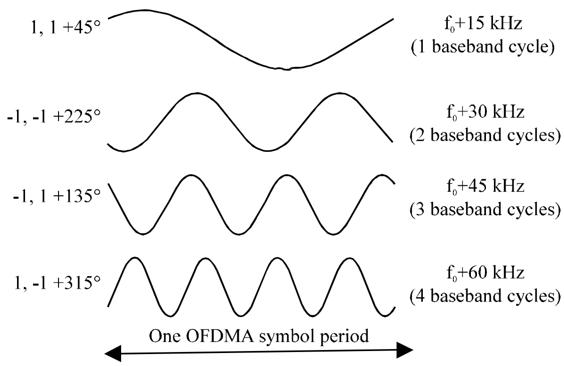


Fig. 9: OFDM modulates QPSK

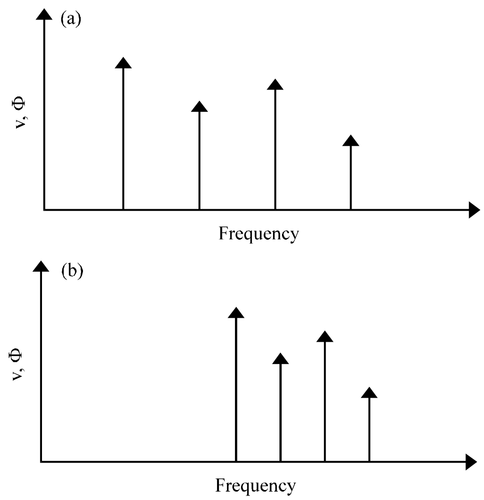


Fig. 10(a-b): SC-FDMA shift allocations

UL and DL have the same frame structure but use different spectra. Rather than transmitting the UL and DL simultaneously in one radio frame slot, TDD transmits only a splitter between DL and UL. This splitter is called a guard period.

In addition, the TDD switch time assigns UL or DL issues to a flexible stretch for UL and DL use (Rumney, 2013; Tran *et al*., 2012).

MIMO technology: The input and output inside MIMO refers to the channel of transmission to the receiver. Multiple antenna techniques provide increased coverage and physical layer capacity. Three types of multiple antenna categories are applied. First, path diversity improves system performance. Thus, diversity can work on the transmitter or receiver or both simultaneously. Second, beam steering to improve coverage consists of little cells that cover a given area. Third, spatial multiplexing enables the use of separate antennas in space to allow simultaneous transmission of more than one stream of data in both time and frequency. In addition, beam forming is a more complex technique used to further enhance transmission. As such, the feedback of the channel condition is used to pre code the signal (Li *et al*., 2010).

When used, the input and output terms refer to channels not antennas. MIMO technology itself has many different types. Single Input Single Output (SISO) which is the most basic radio channel access mode, is defined as a single antenna in the transmitter and a single antenna in the receiver. Single Input Multiple Outputs (SIMO) is a diverse receiver that has one transmitter and two receivers. Unlike SISO, SIMO has low signal to noise ratio. A low signal to noise ratio results in a more robust radio signal (Lee *et al*., 2009). Multiple Inputs Single Output (MISO) is a transmitted diversity technique with a single receiver antenna. Thus, MISO is similar to SIMO but the former does not increase data rate unlike the latter. MIMO transmitter diversity and receiver diversity can increase spectrum efficiency and channel capacity for different signals sent by various transmitters that represent different data streams, each of which is sent at the same time and frequency (Dahlman *et al*., 2011).

Different data streams have been received by each antenna in receiver diversity. Each data stream is decoded and user data are retrieved. This process is called spatial multiplexing, with the number of antennas in MIMO exceeding more than two.

Having more than one antenna does not mean that the system is MIMO. SIMO plus MISO does not equal MIMO because both SIMO and MISO use a single data stream, whereas MIMO uses various data streams (Rumney, 2013).

Diversity can be merged with MIMO spatial multiplexing to improve performance. With regard to MIMO, the path has to be further de-correlated. Thus, the layer with spatial multiplexing becomes synonymous with the stream.

How does MIMO work? Both sides of the sender and receiver which use various data streams, may use more than one antenna to ensure that more than one channel is implemented in modulating the data stream at the same time and frequency.

MIMO benefits from the multipath feature and uses multiple antennas to transmit multiple signals in parallel (Li *et al*., 2010).

In residential areas, these signals are disrupted by many things, such as buildings and trees. Signals maintain their path to the receiver but become affected by disruptions from different locations. Two signals containing different user data are simultaneously transmitted at the same time and frequency. The multipath feature shows that diverse signals reach the receiver at various times, even though these signals have been sent at the same time (Kishiyama *et al*., 2013). The final destination at the receiver will apply a special signal processing algorithm to decode the multiple signals and transmit it as one signal or to select the optimum signal that has been transmitted originally by another. Therefore, a couple of diversity transmitted signals are decoded through one or more antennas (Lee *et al*., 2009).

By using a single channel, many data streams are transmitted at the same time. Thus, multiple antennas collect diverse multipath signals.

LTE-A is set by using QPSK, with 16-QAM and 64-QAM as modulation formats selected based on specific radio channel conditions. A part of LTE-A uses MIMO technology to improve data capacity. MIMO technology is divided into DL and UL, with DL engaging in TX diversity, RX diversity and single-user MIMO beam forming (Li *et al*., 2010) (Fig. 11).

Related to UL in MIMO technology occupied by multiuser MIMO, LTE-A reached 300 Mbps (4×4 MIMO, 20 MHz, 64-QAM) as its peak data rate. This data rate is counted as DL and 75 Mbps (20 MHz, 64-QAM) is counted as UL (Rumney, 2013).

The bearer service LTE-A carries out only packet switch and not circuit switch; thus, Voice over Internet Protocol (VoIP) must be used for voice communication. This application requires priority handling or some level of quality service implementation (Dahlman *et al*., 2011).

LTE-A ARCHITECTURE

System Architecture Evolution (SAE) indicates the overall system architecture for LTE-A, including parts with their own functionality.

Figure 12 presents the LTE-A basic contents. Three nodes of evaluation node B (eNB) form the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) which connects with the Evolved Packet Core (EPC) part of the Mobile Management Entity (MME) Serving Gateway (S-GW)/Packet Gateway (P-GW) data.

The base stations (eNB) are directly connected to the core via one interface. The base station is interconnected to each other through two interfaces which are used to prepare forward packets during handover situations. The right figure explains in detail the functional split between a base station (eNB) and EPC, consisting of MME, S-GW and P-GW. The base station constitutes E-UTRAN and an enabling radio access network which means that many tasks are performed through the protocols (Fig. 12) (Rumney, 2013; Chen and Wang, 2010; Ghosh *et al*., 2010).

The radio access network physical layer has a Media Access Control (MAC) layer that maps the logical and transport channel demultiplexing scheduling information. This information reports hybrid Automatic Repeat Request (ARQ) which is a priority handling and transport format selection.

The Radio Link Control (RLC) layer performs re-segmentation, concatenation, in-sequence delivery, duplicate detection, Service Data Unit (SDU) discard and reestablishment (also Acknowledge Mode (AM), Transparent Mode (TM), Unacknowledged Mode (UM) and Automatic Repeat reQuest (ARQ). The packet data convergence protocol layer works on robust header

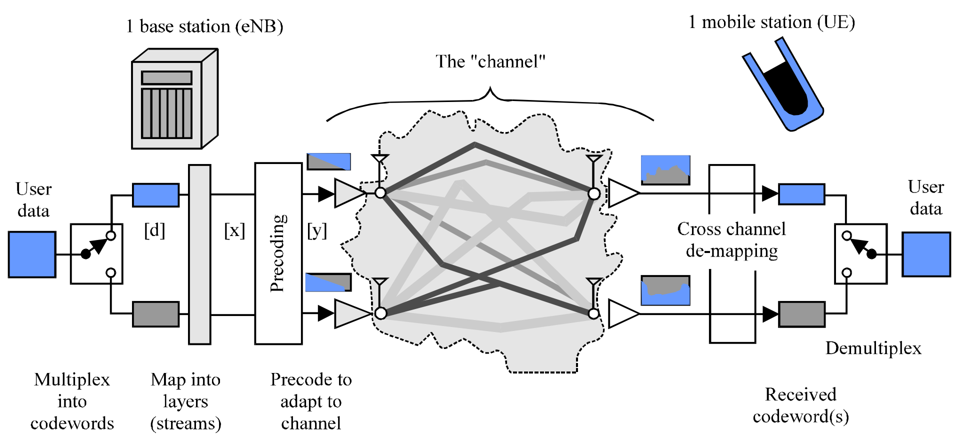


Fig. 11: Simple MIMO architecture

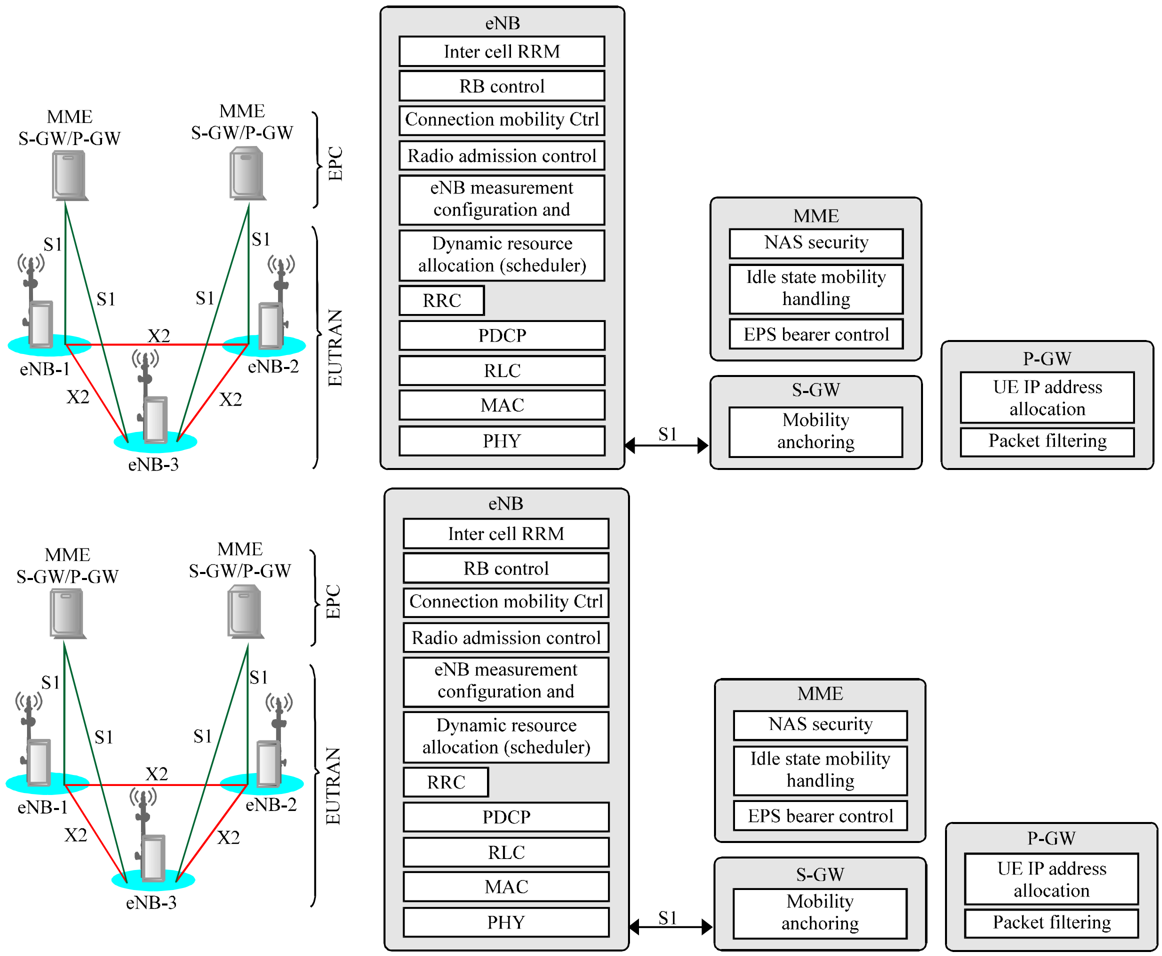


Fig. 12: LTE-A SAE

compression in the sequence delivery of upper layer Protocol Data Unit (PDU) and duplicate elimination of lower layer SDU ciphering of the user. Aside from protecting the control plan integrity of control planes by using the time-based discard method, the efficiency of transmission to the layer of the protocol stick called the user plane should be improved. The last layer is a Radio Resource Control (RRC) layer that works on broadcast paging, connection setup, radio bearer control, mobility functions and measurement control in terms of function. This layer shows how the base station works (Ghosh *et al*., 2010).

The main security task of the base station interconnected with the MME in EPC Non-Access Stratum (NAS) is Evolved Packet System (EPS) career management authentication. The other tasks of this station include mobility handling, paging origination in EPS connection management security control, idle state mobility handling and EPS.

The User Equipment (UE) connects with the MME layer through the base station. This level is called the control plane, in which the NAS layer in UE interconnects with the NAS layer in the MME.

The control plane broadcasts system information which initiates paging over the interface and sets up the connection in the RRC level that controls all radio carriers over the interface. The S-GW acts as a mobility anchor and packet data network gateway (Dahlman *et al*., 2011; Madan *et al*., 2010).

LTE-A is differentiated by the following multiple points:

- Enhanced execution at cell edges such as DL 4×4 or 8×8 MIMO at least 2.4 bps/Hz/cell
- Increased amount of active subcarriers that work simultaneously
- Growth summit data rate, UL 1.5 Gbps and DL 3 Gbps

Table 2: Brief review of LTE-A

	LTE 3GPP release 8	LTE-advanced 3 GPP release 10	IMT-advanced International telecommunications union "true 4 G"
Peak (DL)	300 Mbpps	3 Gbps	100 Mbps (high mobility)
Data rate (UL)	75 Mbps	1.5 Gbps	1 Gbps (low mobility)
Peak spectrum (DL)	15	30	15
Efficiency [bps Hz^{-1}] (UL)	3.75	15	6.75
Tx bandwidth (UL and DL)	Upto 20 MHz	Upto 100 MHz	Upto 40 MHz

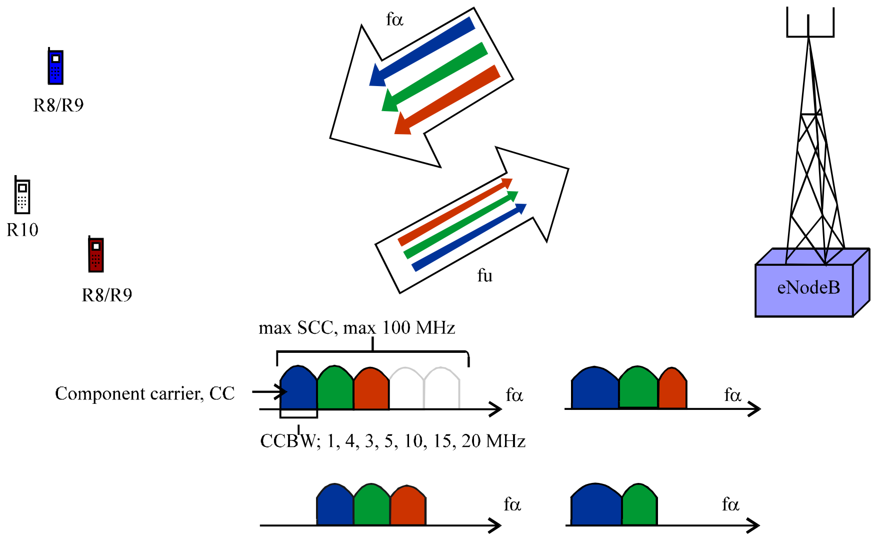


Fig. 13: FDD CC aggregation

- Imposed spectral performance of 30 bps Hz^{-1} compared with a maximum of 16 bps Hz^{-1} in Release 8
- The main key launched by LTE-A is CA which improves the use of multi-antenna technology and introduces the functionality RN (Table 2)

Carrier aggregation: The extended maximum transmission bandwidth is a plain expanding upto 100 MHz by aggregation. As such, LTE-A is still compatible in reverse with older versions such as R8 and R9.

By aggregating five LTE 20 MHz carriers known as Component Carriers (CC), the lack of sufficient contagious spectrum forces the use of CA to meet peak data rate targets.

The tangible points achieve wide bandwidth transmissions and facilitate efficient use of fragmented spectrum and efficient interference management for control channels in heterogeneous networks. Note that CA can significantly work over FDD or TDD (Chen and Wang, 2010; Ghosh *et al*., 2010; Ratasuk *et al*., 2010).

The number of aggregation carriers varies for the UL and DL. Moreover, the number of UL CC cannot be larger than that of the DL CC. CC may individually vary based on bandwidth numbers. Figure 13 shows the FDD CC aggregation (Fig. 13) (Rumney, 2013).

The eNB in the LTE-A or R10 can receive and allocate up to five CC for DL and UL, whereas the eNB in R9 and R8 can work with only one CC.

By using contiguously arranged CC aggregations, this method considers the simplest way when including the same working frequency band which is called intra-band contiguous. Intra-band contiguous does not always occur given the location of the frequency (De La Roche *et al*., 2012).

In intra-band noncontiguous, a CC belongs to the same band but is isolated by frequency space. In inter-band noncontiguous, a CC belongs to different frequencies and works in a different frequency band. Figure 14 shows these three types of band.

The serving cells deployed into each CA are concomitant with each CC. Thus, the coverage may diverge according to CC frequency. Power planning is considered a tangible solution for the heterogeneous network strategy.

The RRC connection is managed by one cell plus the primary serving cell rendered by the primary components

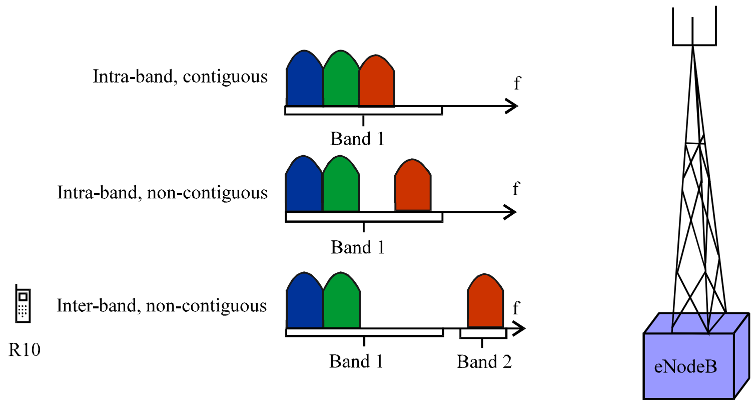


Fig. 14: Inter-band and intra-band CC aggregation

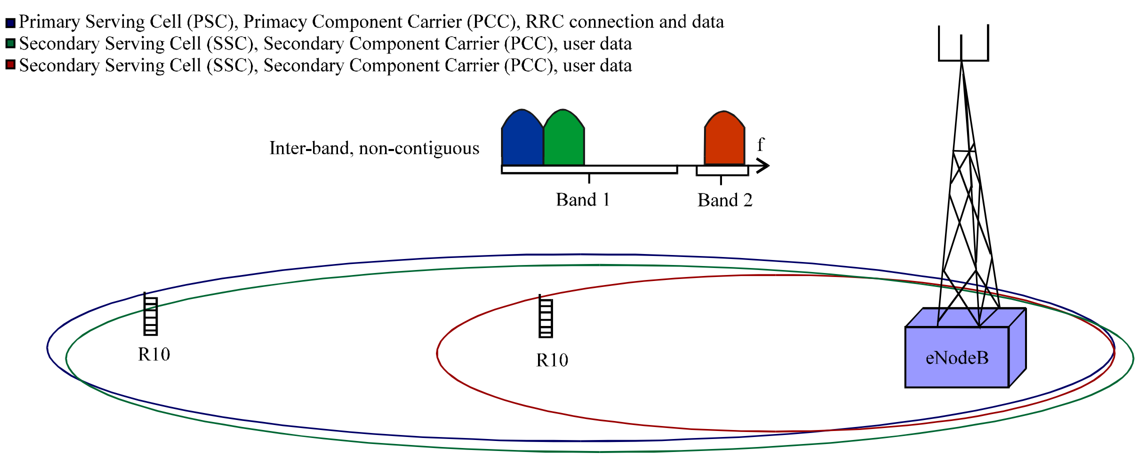


Fig. 15: Serving cell carrier aggregation

carrier which includes both DL and UL. Other carriers include the secondary components carrier which includes UL and DL and employs the secondary serving cell.

Many barriers can be designed to supply various degrees of coverage. The inter-band CA of the CC may suffer from considerable path loss which increases growth frequency. Figure 15 shows the three different types of CC that can be launched only by black UE. The white UE does not fall under the coverage area of red CC (Chen and Wang, 2010; Tran *et al*., 2012).

Various coverage areas have different serving cells. Moreover, each CC is identical to the serving cell. The MAC and PHY protocol layers are most affected by the presenting CA. However, some recent RRC messages are also inserted.

The LTE-A work is restricted within the MAC and PHY layers when completing the pyramid IP layers, such as the application layer and transport layer.

A stick of the IP suite contains a set of layers. Each layer contains a number of functions when integrated into the pyramid of communication stick (Chen and Wang, 2010; Ratasuk *et al*., 2010).

Relay node: LTE-A deploys its own coverage methods through inner band and outer band relaying. This feature considers differences in LTE. This new feature was developed recently and included in LTE-A as well.

To enable qualified heterogeneous networks to obtain self-backhaul of the connection link between UEs and eNB or the relay base station, the RN acts as the eNB in low power mode to guarantee coverage and suitable capacity at cell edge. This technology supports LTE-A to expand coverage to reach specific areas in low power mode which equates to low cost.

The RN is connected to the eNB to produce the concise form called GReNB. Through the radio interface, RN also linked to the UE interface at another side.

Therefore, the radio resources will be shared among GReNB, RN and UE. The GReNB can serve its own UE which converged at its own signal.

The RN provides the same GReNB that is employed to terminate the radio protocols of the UE interface. Moreover, the GReNB provides the RN a subset of the UE functionality.

RN prevails in the cell by using the individual ID of the physical cell in addition to supporting channel synchronization. Therefore, all UEs are completely under the control plane and the user plane protocols are created in the RN. Figure 16 shows the RN operations.

The two types of RNs are as follows:

Type 1: Non-transparency relaying which supports the far off UE located away from the GReNB. Consequently, this type needs to send a general reference signal to detect the GReNB information. As such, the main purpose of type 1 is to provide permanent coverage which leads to expanded communication services and isolated UE data transmission service connections

Type 2: Transparency relaying which supports the local UE within the GReNB coverage. The local UE also had another connection to the GReNB through direct connection with the RN linked to the GReNB. Moreover, type 2 supports boosting of the link connection to improve the Quality of Services (QoS). This type is unable to send reference signal and conduct information checking control but is able to support the space amplification of the network by employing multipath-converged local UE

TRANSPORT LAYER

The data exchange from sender to receiver and vice versa is the main obligation of the transport layer. This data exchange needs to be accurate from end to end. Some applications require transport in reliable environments. Moreover, another application is needed to move data inside secure channels and some applications demand fast data relocation.

The transportation process needs a flow control to ensure that packets have been switched correctly. Inside the transport layer, many protocols work with each other to operate specific application needs (Kurose and Ross, 2013; Peterson and Davie, 2007).

Role of the transport layer: In higher layers, the transport layer supports connection-oriented or connectionless communication and prioritizes normal data units to be transferred. The transport layer allows a network to transport only small data units, wherein layer 4 can be divided by layer 5. In the session layer, incoming large data packets are transported into the small PDU via segmentation, send over the network and transferred into the opposite transport layer of the target system. Subsequently, the large data packets are reassembled to their original form (Kurose and Ross, 2013).

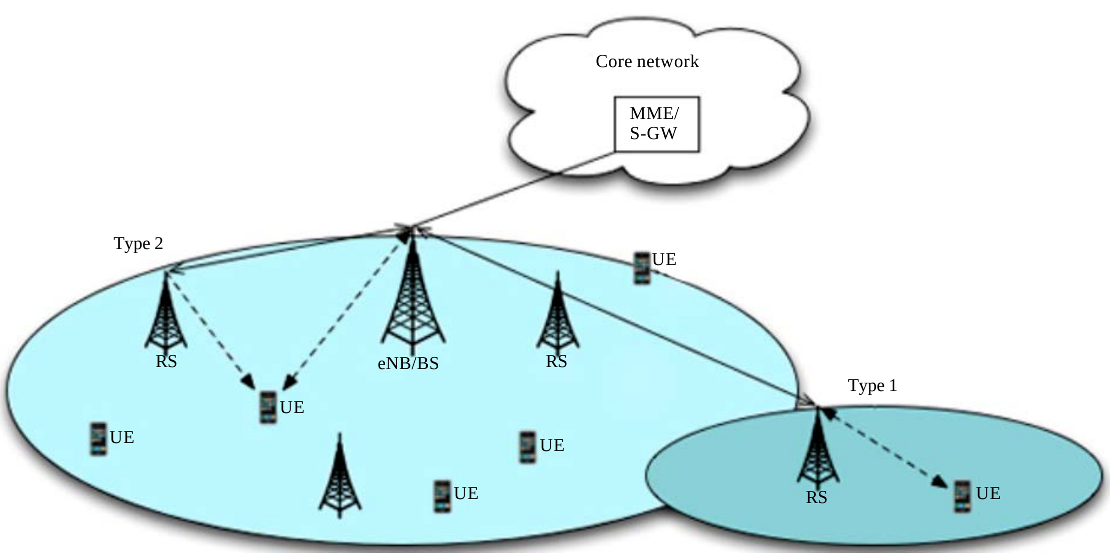


Fig. 16: Relaying node in LTE-A

The small data units of layer 5 are transported through the network. This transfer would be ineffective for the station because of the "concatenate" function. Concatenation can form a larger unit that transports them through the network. The remote site "separates" (separation) these data units back to their original small units (Kurose and Ross, 2013).

Different classes of transport layer service: The transport layer supports the selection of QoS parameters. In the OSI transport protocols, the transport classes have five classes (service 0-4) set with different performance characteristics (Peterson and Davie, 2007). The transport layer can choose a specific application or relationship between pluralities of paths to the data units because of data flow over multiple paths to a destination.

For connection-oriented communication, the transport layer assigns the correct sequence number to restored data units (Peterson and Davie, 2007). The transport layer expands end-system connections, specifically, from end system to end system to user connections which is from user to user.

Many protocols work on the transport layer platform, with all of these protocols receiving data from application layers or packets from the network layer. These protocols supply special requirements for specific application demands, including video streaming which is served by the User Datagram Protocol and web browsing traffic application which usually requires TCP. Additionally, new protocols, such as the Stream Control Transmission Protocol (SCTP), take advantage of both UDP and TCP. SCTP offers new features and differs from UDP and TCP.

UDP: UDP is one of the main protocols commonly used inside IP stick applications that require fast transform regardless of reliability. UDP is one of the protocols that work inside the transport layer. The UDP header is smaller than TCP and SCTP and does not use handshake conversation because the UDP header is not designed for reliable application in the UDP which is known as connectionless and disoriented. Therefore, UDP is suitable for applications that do not need to undergo error detection or correction process. The UDP header size is smaller than that of TCP because the former does not contain the acknowledge feature and error detection function.

TCP: TCP is one of the most vastly utilized protocols in the transport layer and Internet suite protocols. TCP is coined as the connection-oriented protocol given its numerous applications. TCP is implemented in e-mails, web browsing, data exchanges and growth fraction of multimedia components transportation in real time over web traffic Hypertext Transfer Protocol/TCP. In addition, the transfer process requires a real-time secure environment over Hypertext Transfer Protocol Secure/TCP.

The segment of TCP is the PDU inside the transport layer transmission. By adding the TCP header to the PDU, the header size equals 32 bits. The TCP header consists of the source and destination port address, sequence number, acknowledgment (ACK) number, checksum, receive window and flag field. Figure 17 shows the contents of the TCP header (Kurose and Ross, 2013).

Sending and receiving among terminals is determined by many criteria, such as bandwidth size which links these terminals, media connection type in the physical layer, distance between sender to receiver over which media is accessed and the number of terminals that use bandwidth. The data flow control utilizes carrier bandwidth according to the number of terminals that send and receive

Offsets	Octet	0								1								2								3							
Octet	Bit	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
0	0	Source port																Destination port															
4	32	Sequence No.																															
8	64	Acknowledgment No. (if ACK set)																															
12	96	Data offset				Reserved 0 0 0			NS	CWR	ECE	URG	ACK	PSH	RST	SYN	FIN	Window size															
16	128	Checksum																Urgent pointer (if URG set)															
20 ...	160 ...	Options (if data offset>5. Padded at the end with "0" bytes if necessary) ...																															

Fig. 17: TCP header

bandwidth. Considering the data size, the utility increases in proportion inclusive of the number of sender and receiver in the transport area.

Many aspects are related to the data transfer process in TCP, such as reliable transmission, error detection, flow control and congestion control. Each of these four aspects comprises the main functionality of TCP. These aspects are briefly described as follows.

Reliable transmission refers to deployment of an identical method that identifies each data content exchange by using a sequence number. However, reliable transmission implements the three-way handshake to settle the above process.

Error detection employs a sequence number that grants a receiver the ability to transport a duplication packet. Moreover, correct sequence rehash packets are sent to distinguish lost packets by using ACK for more accurate holding checksum (Peterson and Davie, 2007).

Flow control runs the end-to-end flow control protocol to prevent the sender from sending data to the receiver too fast. Flow control ensures that receivers obtain the packets reliably and correctly.

Congestion control TCP has a set of techniques and methods launched to gain high rendering and to avoid collision, whereas network performance can collapse when affected by many factors. Therefore, these methods dominate the rate at which data are transported in the network. Likewise, the decrease in data flow rate would induce breakdown; thus, distribution among flows should be adjusted.

Many studies on TCP over LTE and LTE-A have presented various solutions. These solutions are discussed below.

This study has intensively reviewed and analyzed many previous studies related to TCP over LTE/LTE-A (Table 3).

The performance of TCP over LTE presents some aspects comparable to TCP auto-tuning. Some significant elements, such as TCP window size, loss probability and Round Trip Transmission (RTT), are available in TCP execution in contrast to wireless medium which might present new challenges.

Moreover, TCP cubic and Windows auto-tuning are not custom-made for the WiMAX link for connections overpass, even though WiMAX belongs to 4G technology like LTE-A:

- The results of TCP auto-tuning over AM LTE scored approximately 197 Mbps with loss probability of 0.001%. In addition, the packet lost percentage in the wireless last hop between UE and eNB reached 0% when supported by the RLC ARQ. Interestingly, the results, especially the throughput factor, suffer from the outcomes affected by various Operating Systems (OS) because this approach is not suitable for Windows XP because TCP/Ipv6 protocols do not support Windows scaling. This finding is expected given that the transport layer can support any OS because the original protocol is compatible with any OS. Moreover, auto-tuning Windows is not suggested for WiMAX traversing connections. In addition, the out-of-sequence delivery technique is not included in LTE properties which require packet reordering. Packet reordering is not infrequent in the present high-speed network (Park *et al*., 2011)
- The new congestion control algorithm named TCP-Future Internet Technology institute (FIT) accomplished impressive results over a real environment by considering two factors, delay and packet loss. Both factors have been recruited as inputs inside the hybrid congestion control which targets applications. The main enhancement of TCP-FIT is the implementation of variable N in parallel virtual sessions. However, only one physical connection is used for these sessions. N is given; however, the time required for each virtual session to manage packet loss needs to be calculated. TCP-FIT inventors did not show how they developed the

Table 3: A review of previous studies

Author	Environment	OS	Description
Hyun Seo Park, Jae Yong Lee and Byung Chul Kim	Real network	Win XP, Win 7, Win 2008, Linux	TCP auto-tuning enabled LTE AM TCP window scaling Out-of-sequence delivery
Jingyuan Wang, Jiangtao Wen, Jun Zhang and Yuxing Han	Real network	Linux, Windows	New TCP congestion control algorithm
Ghassan A. Abed, Mahamod Ismail and Kasmiran Jumari	Simulation	NS2	New TCP congestion control algorithm Measured and analyzed various TCP congestion control factors over different networks
Peng Shang, Yuhui Zeng, Jinsong Wu and Pei Xiao	Simulation	Not reported	TCP mapping through static table and priority selection and channel selection New radio resource management and enhanced QoS
Ke Liu and Jack Y.B. Lee	Real network	Linux, Windows	Mobile data network Mobile accelerator optimization Enhanced TCP performances

mathematical formula adopted for slow start, congestion avoidance, fast transmit and fast recovery. Therefore, subsequent researchers may focus on this issue in the future to shed light on some of the points presented, such as the method of targeting applications and the file size of the data transferred. Moreover, the results come from one sender to one receiver which is a condition ideal for the network (Wang *et al*., 2010)

- The comprehensive measurement and analysis of various types of TCP behavior over different media carriers presented the desired results through different simulation scenarios which determined differentiation among TCP protocol. The enhanced TCP congestion control algorithm has considered the following four factors: Slow start, congestion avoidance, fast recovery and fast retransmit. Some experimental documentation did not include time of simulation period which was corrected later. In addition, the main bandwidth that linked nodes in some simulation scenarios seems capable of including all nodes connected to it. The number of nodes was supposed to be less than or equal to the total bandwidth to clarify congestion control when faced with real events (Abed *et al*., 2011a, b, 2012)
- The novel resource scheduling provides enhanced QoS across radio resource management by using a selective channel aside from mapping the TCP which is selected from the static table. This approach leads to TCP with reduced RTT, evolved efficiency of TCP ACK and more stable performance. The percentage of enhanced FTP UL and DL throughput was approximately 30%. The delay time also improved by approximately 3-5 sec. The delay time of VoIP enhanced for the delays of ACK packets was approximately 20 msec. Despite the implementation of the new technique, the results seem somehow encouraging. However, this technique has the following limitations: the measurement results included two users only; the file size throughout the experiment was not mentioned; no workload is present inside the three connection types because no drop was observed regarding the congestion event; the technique seems focused on QoS because the novel approach worked on the data link layer; optimization in the data link layer slightly affected this management; the simulation environment was not described (Shang *et al*., 2013)
- The mobile data network acted as a new method working inside the network layer at the mobile client side by using mobile accelerator notation. This notation acts as external gate or external interface for the network layer to classify the packet and then accelerate the TCP. This approach presented enhanced TCP performance by 200% which is considered a sanguine progress. However, the term optimization in mobile acceleration suffers from many limitations. For instance, implementing the packet classifier at the mobile client side will increase the process time. The target layer is the network layer that has been added to the mobile data network despite being illustrated as not needing any implementations in the server or client. The TCP acceleration inside the mobile data network looks vague when implemented by the mobile client (Liu and Lee, 2011)

This study has reviewed many issues such as enhanced performances regarding the handover or developed cross layer. The progressive results obtained for QoS were impressive which may positively affect the corresponding TCP performance. Furthermore, implementing TDD over LTE can improve the benefits of TCP (Crosnier *et al*., 2012; Pacifico *et al*., 2009; Susitaival *et al*., 2010; Zhou *et al*., 2013; Chung *et al*., 2010).

Consequently, although TCP has been receiving many enhancements through the adoption of the congestion control algorithm or the supplementation of new techniques, the TCP suffers many problems inside the internet infrastructure or inside its own characteristics. These problems are accompanied by tremendous increase in the number of end users and presence of a new high-performance mobile technology (Caro *et al*., 2003). Given the urgent demands inside the transport layer to operate a new protocol capable of exceeding the limitations of TCP, this study presents SCTP.

SCTP: SCTP is another protocol in the transport layer next to TCP and UDP. SCTP combines the properties of TCP and UDP and is message-oriented and reliable. However, SCTP relies on packets or datagrams and not on byte counts as TCP does. The SCTP was created to enable signal transmission to IP telephony (Stewart and Xie, 2001).

SCTP implements the concept of an association wherein the connection is established to transport multiple message sequence data streams located and sustained with each other but potentially following/preserving more non-numbers.

Furthermore, SCTP converted data received from the application layer into portions of streams encapsulated into streams in chunks after adding the SCTP header to move the SCTP PDU to the next layer.

Bits	0-7	8-15	16-23	24-31
+0	Source port		Destination port	
32	Verifcation tag			
64	Checksum			
96	Chunk 1 type	Chunk 1 flags	Chunk 1 length	
128	Chunk 1 data			
...	...			
...	Chunk N type	Chunk N flags	Chunk N length	
...	Chunk N data			

Fig. 18: SCTP header

The SCTP header has 32 bits of checksum. In addition, the verification tag has two functions that support recognition of an association from end to end of the same pair connections and guarantee security which was not included in the previous version of SCTP PDU (Cui *et al*., 2011; Stergiou *et al*., 2004). The checksum provides the correct transportation process and accrues data integrity. Figure 18 shows the SCTP header (Stewart and Xie, 2001).

SCTP implements 32 bits for the checksum instead of 16 bits in TCP and use multiple streams. SCTP introduces a chunk of stream as a series of messages and not as byte stream as in TCP. As such, the header, together with some chunks that are controllable and movable, will be commonly added to PDU. The SCTP presents a four-way handshake to establish an association connection and employs a three-way handshake to shut down. Additionally, the SCTP does not provide a semi-open connection (Caro *et al*., 2003; Dreibholz *et al*., 2012).

SCTP provides two notations, namely, multi-homing and multi-streaming. Multi-homing SCTP offers end-to-end communications ability through various media, such as Ethernet and Wi-Fi, even though each association uses its own IP pool. Multi-streaming provides many unidirectional physical streams or channels between sender and receiver or end-to-end terminals in association (Stewart and Xie, 2001; Dreibholz *et al*., 2012; Kashwan and Karthik, 2012; Xu *et al*., 2013).

RESULTS AND DISCUSSION

This study validates the systematic SCTP model addition to TCP over LTE-A by implementing two simulation scenarios by using NS-2.

Both scenarios show how the protocols perform during exposure workloads through a transmission process and point out that all of these protocols are relabeled, with each of them sending ACK to the sender. As such, the fully used single path will suffer because of high intensity exchanges through senders and receivers and because of returned ACK to the senders.

First scenario: The first systematic scenario has been executed through 10 UE nodes linked to the server via two nodes. These nodes are connected to the gateway. Two groups are observed in each side. Group 1 contains five UE1 nodes that link to the eNB1 node. Group 2 contains five UE2 nodes that link to the eNB2 node. Both eNB1 and eNB2 are connected to the gateway; thus, gateway 2 is attached to the server, as shown in Fig. 19.

One of these UE nodes in each of two groups used SCTP, whereas the rest of the nodes used various types of TCP, such as Reno, Vegas and the standard TCP.

This study tries to clarify the facts. A large difference is observed between networks in reality and what were implemented in the previous studies, whether those studies used either real network experiments or simulation experiments.

The number of selected sample of end devices that send and receive over the network is low in comparison with networks in reality. Furthermore, the bandwidth link among the samples is almost ideal which means that no workloads are carried out. We will clarify the ideal performance and the performance under workload circumstances in the following text.

For the first scenario with 10 nodes over the high-speed wireless environment, the bandwidth suffers from the workload that is, the number of UE bandwidths are greater than or equal to the bandwidth connection between the gateway and server which is synchronized with delay time.

This study included a set of performance factors inside its protocols. These factors aim to show how these protocols work. Therefore, the determined process for each protocol will be measured and analyzed.

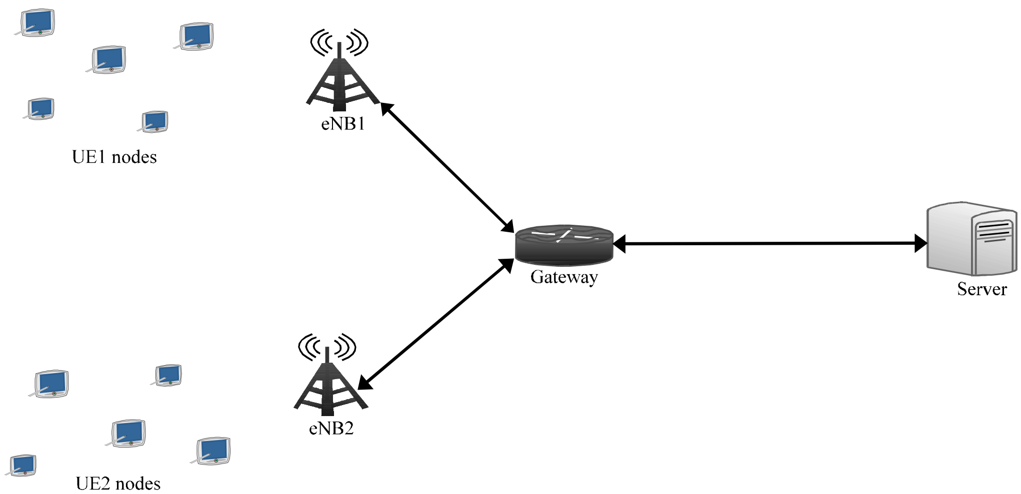


Fig. 19: First scenario SCTP and various TCP topologies

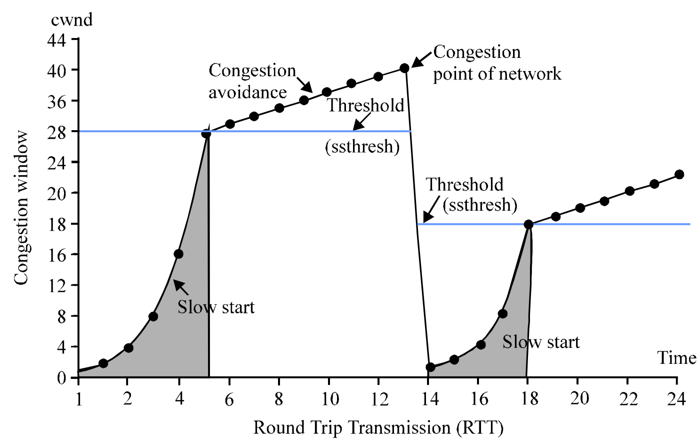


Fig. 20: Slow start and congestion avoidance

The performance factors which have been considered as the congestion window (cwnd), is a factor that set forth the number of bytes than can be pending at any time. The factor cwnd uses a number of segments that passed successfully at each time within a specific period.

Furthermore, the window size is preserved by the receiver. Thus, when workload overload occurs, the linked line between two places stops being overloaded. The window size is computed by determining how large the congestion between sender and receiver devices is.

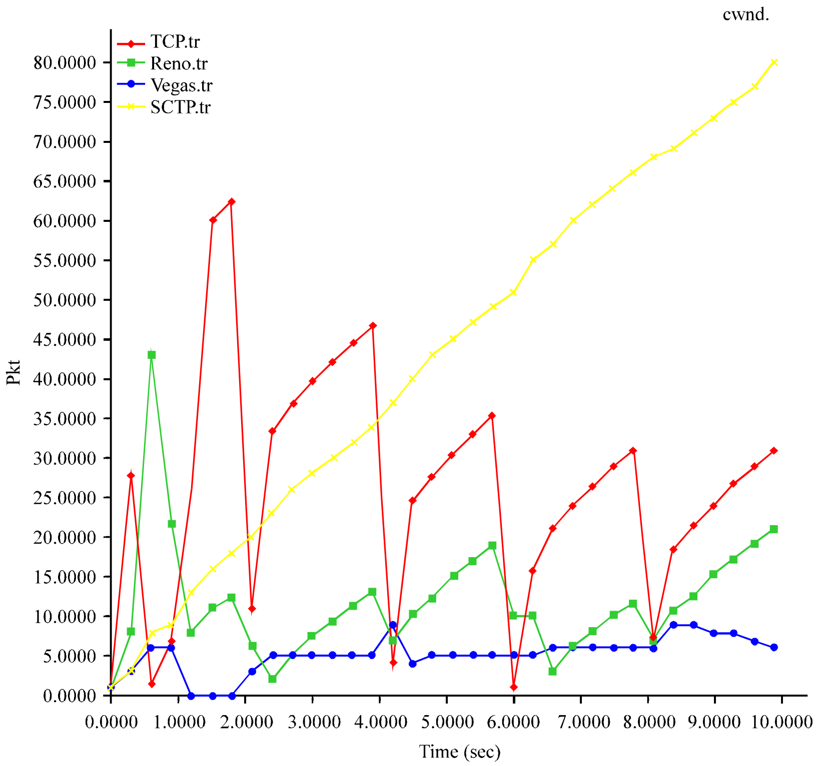
The factor cwnd captures the congestion control phases, such as slow start, congestion avoidance, fast recovery and fast transmit.

Slow start is the amount of increase after each successful acknowledgment (ACK) until reaching a phase, usually cwnd = 1 at the beginning and then increases to cwnd = cwnd+1 until reaching the threshold level called congestion avoidance because congestion level in the network will appear. Inside the slow start phase, another parameter that is, the slow start threshold (ssthresh), occurs after the congestion point state occurred. Thus, the number of cwnd decreased until the network returns to the normal congestion point. Figure 20 shows the RTT.

Consequently, Fig. 21 and 23 present the results of plotting cwnd of SCTP in comparison with various TCP over LTE-A and wire carriers, respectively.

The first scenario, shown in Fig. 21, illustrates the cwnd of SCTP compared with the cwnd of various types of TCP protocols which describe the successful exchange segments after obtaining the ACK.

Disparity regarding performance behavior for each protocol is observed in the results of the plotted cwnd (Fig. 21). The results provide the first scenario of the



Parameters	Value
Transport protocol	TCP, Reno, Vegas, SCTP
Propagation delay of all nodes	2 MS
Antenna	MIMO 2*2
Distance Far UE	1~2 km
Bandwidth of eNB link	50 Mbps
Bandwidth of UE's link	10 Mbps
No. of UE nodes	10, 5 on each eNB side.
Bandwidth of server link	100 Mbps
Packet size	1500, 2000 and 2500 byte
Window size	

Fig. 21: A compression cwnd of SCTP and various TCP over LTE-A

systematic model with a time period of 10 sec and the highest PDU of 80. The SCTP which appears in yellow, ranked the highest among other TCP protocols. The SCTP plotted slope reached the peak by achieving 80 which is the highest level of segments exchange.

A step deeper at 1.5 sec, the original TCP protocol (exhibited in red) obtained 60 PDU which equals the TCP reached the point of congestion. Therefore, a drop after congestion avoidance recall was observed on the basis of ssthresh = cwnd/2, whereas the SCTP progressed continuously upward. The second 2.1 SCTP attained 25 PDU compared with TCP which scored 10 PDU.

This trend continues in this vein until the time period is finished. Figure 21 shows the plotted cwnd which presents other types of TCP, such as Reno (green) and Vegas (blue) which were worse than TCP.

Other TCPs were unsatisfactory compared with SCTP which utilizes more than the rest of the TCPs by scoring high PDU. The increase of SCTP was slow but still durable. We are aware that all protocols inside this scenario were overloaded.

Second scenario: The second scenario used two bottlenecks between the client and server entities. The two bottlenecks are represented by two gateways: Gateway 1 and 2. Five clients are connected to gateway 1. Gateway 1 links to 2 which in turn is connected to five servers.

As such, the second systematic scenario has two bottlenecks that work over the wire media. Through this scenario, five clients are connected to gateway 1; gateway 1 is in turn connected to gateway 2 through a single path; gateway 2 is linked to five servers as a final destination.

One client uses the SCTP. On the other side of the two bottlenecks, a server received SCTP traffic. Other clients utilized several types of TCP, such as New Reno, Reno, FACK and Vegas. On the other side of the second gateway, the servers received data from clients, each of which is based on the equivalent type through one path between two bottleneck routers.

Related to the second scenario, Fig. 22 shows the topology of this scenario.

The second scenario model presents another shape of the workload of the systematic model that emerges from real circumstances in the network. Therefore, the second model has been selected as a sample test exchange segment through two bottlenecks by using only one single path to link the client side to the equivalent server side. Moreover, the main path gateway's

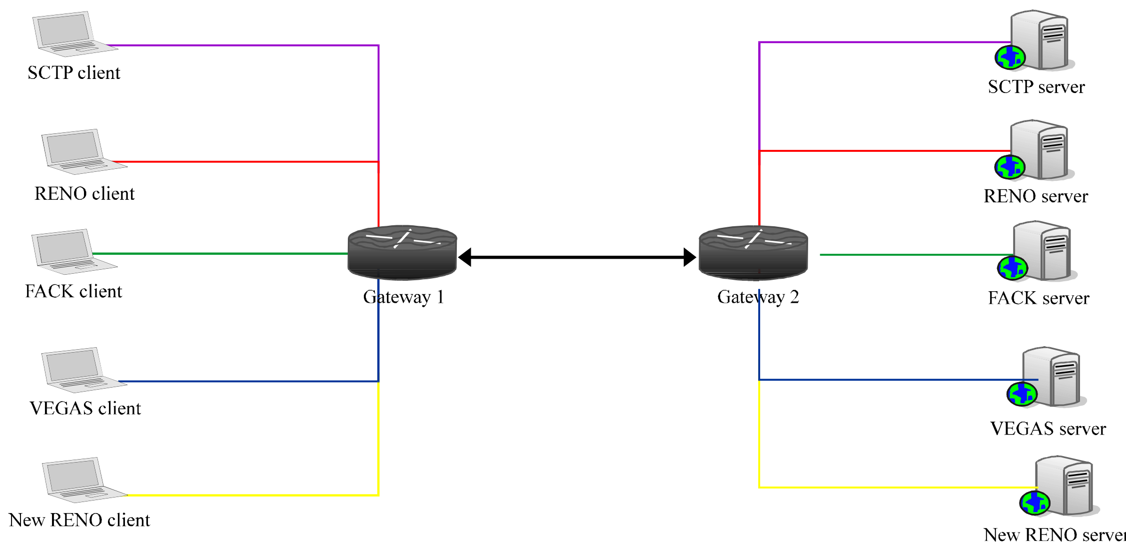
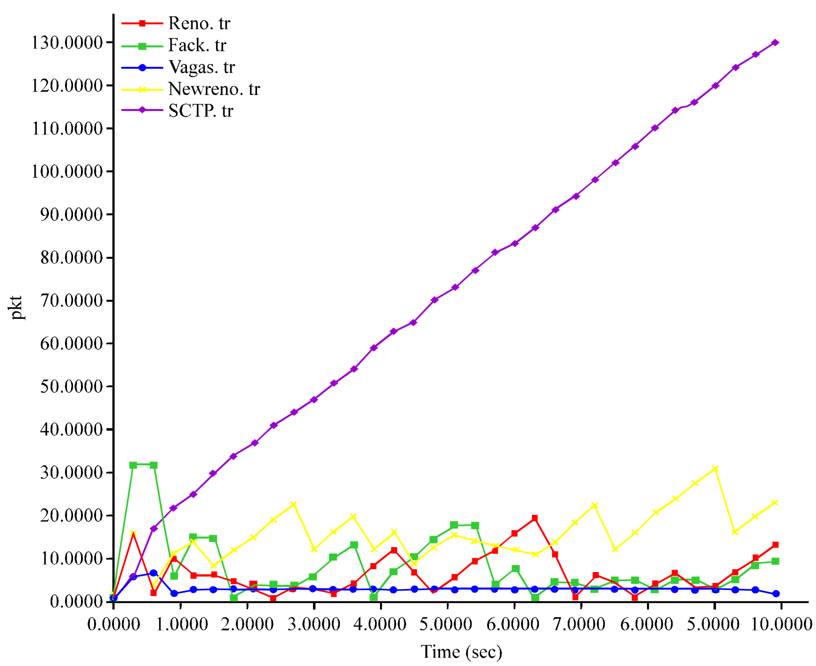


Fig. 22: Second scenario SCTP and various TCP topologies



Prameters	Value
Transport protocol	Reno, FACK, vegas, NReno, SCTP
Propagation delay of all nodes	2 Ms, 3 Ms
Bandwidth of node links	20 Mbps
Bandwidth of server link	100 Mbps
Packet size	1,500, 2,000 and 2,500 bytes
Window size	128

Fig. 23: Compression cwnd of SCTP and various TCP

bandwidth size is less than or equal to the aggregate bandwidth size of clients which means that the bandwidth is almost fully utilized by these aggressive protocols.

The returned ACK to the senders would bounce to the workload state, with the sending and receiving operations of reliable protocols consuming more bandwidth size because of the protocol header, window size, ACK, segment size and file size. Therefore, measurements under this environment will provide a viable solution and show the behavior of these protocols clearly.

The variance in the protocols' cwnd performances can still be observed inside the second scenario. However, this time, the amount of disparity is larger than that in the first scenario, as shown in Fig. 23. The results of the performance factor cwnd are presented by the plotted shape which denote that SCTP again overcomes the rest of its competitors.

The second scenario model has a time period of 10.000 sec and, even with an increase in time to 20.000, 30.000 and even 50.000 sec, the results regarding the protocols were exactly same.

The second scenario is precisely the same as the first systematic model, with both scenarios at 5.500 sec. No changes in the TCP side means that the cwnd of various TCPs earned higher PDU than that achieved by other TCPs, except for the SCTP which scored the highest level of PDU at the end.

Therefore, with the second scenario, the highest number of PDU was 130.000 because packet loss in the second scenario was less than the first scenario because the second systematic model used wire media. Otherwise, the first systematic model used wireless media wherein packet loss was higher than the second scenario.

Figure 23 shows that the SCTP (in purple) scored the best segment swap level of 130.000 at the end of the experiment. This result is greater than the rest of the protocols.

However, at 0.400 sec, FACK TCP (in green) achieved 30.000 PDU, whereas Reno (in red) and New Reno (in yellow) achieved 18.000 PDU. SCTP obtained a modest PDU below 10.000 in the same manner as the Vegas TCP (in blue).

Consequently, the TCP types reached the congestion point cwnd = ssthresh. For this purpose, Reno and New Reno drop down as a reaction to congestion avoidance, whereas FACK and Vegas extended one step more with no increments, cwnd = ssthresh. Singularly, SCTP continued progressing against the rest of the TCP types. At 1.300 sec, the SCTP reached 25.000 PDU, whereas other TCPs react to congestion avoidance ssthresh = cwnd/2. FACK and New Reno obtained 15.000 PDU, whereas Reno obtained 7.000 PDU and Vegas obtained 3.000 PDU.

The stereo typed performances remain the same for SCTP. SCTP keeps increasing in the stereotype, whereas other TCPs fluctuate based on the RRT status at 30.000 sec.

At 9.000 sec, the level of SCTP still progressed. Therefore, New Reno scored the highest PDU of 30.000, along with FACK.

In sum, the TCP plot shape was unsatisfactory. Otherwise, SCTP was utilized more than the rest of the TCPs on the basis of the high PDU scores. Although, the increase was somehow deliberate, the increase is still durable because both SCTP and TCP are reliable protocols. We noted that all protocols inside this scenario were underloaded.

More survey performance factors: This part covers some of the performance factors captured and analyzed through systematic review models. The performance factors include throughput, packet loss and queue size.

The throughput network refers to the number of successful PDUs transferred between two connected devices and the median rate of the acknowledged delivery of PDU.

Another type of throughput which is determined by bandwidth, is throughput considered subordinate to the bandwidth. As such, the bandwidth determines how fast a device can send data over a single carrier, thus providing a measure of how fast the PDU potentially moves along.

Throughput between two nodes in the link is usually assumed from node F to node T; hence:

$$\text{Throughput} = \frac{\text{No. of bits from F to node T}}{\text{Observation duration}}$$

which means that the total number of received bits in node T is divided by the time period spent on sending these bits. The plotted throughput of SCTP and TCP is shown in Fig. 24.

The time period measurement of the throughput depended on the first scenario was also 10 sec. The throughput of SCTP (in red) and TCP (in green) are plotted in Fig. 24.

The highest level of PDU throughput was 75 PDU. Both SCTP and TCP started increasing; likewise, the slow start event, shown in Fig. 21, for the cwnd of SCTP and TCP both scored 72 PDU for TCP at 0.2 sec, whereas SCTP achieved 73 PDU at 400 sec. A delay of approximately 200 is experienced by the TCP.

Meanwhile, at 0.700 sec, TCP decreased from 72.000-5.000 PDU, whereas the peak of SCTP still increases. At 1.400 sec, SCTP decreased from 73.000-36.000 PDU; the peak of TCP seemed to increase but is interrupted again at 1.600 sec. Subsequently, the PDU decreased to 37.000 at 1.600 sec, reached 20.000 at 5.2.00 sec and reached 30.000 at 7.000 sec.

TCP returned to 72.000 after 7 sec. However, SCTP decreased at 5.200 sec which reached 50.000 PDU and then maintained its previous position until the end.

On the basis of the throughput as performance factor, a disparity between SCTP and TCP was clearly observed. This study states that measuring and/or analyzing the performance factors of the network is important to determine whether the simulation or hardware device corresponds to real situations in the network.

Accordingly, this study used two systematic models to show the shape of the workload because of the observed difference between the simple model, in which

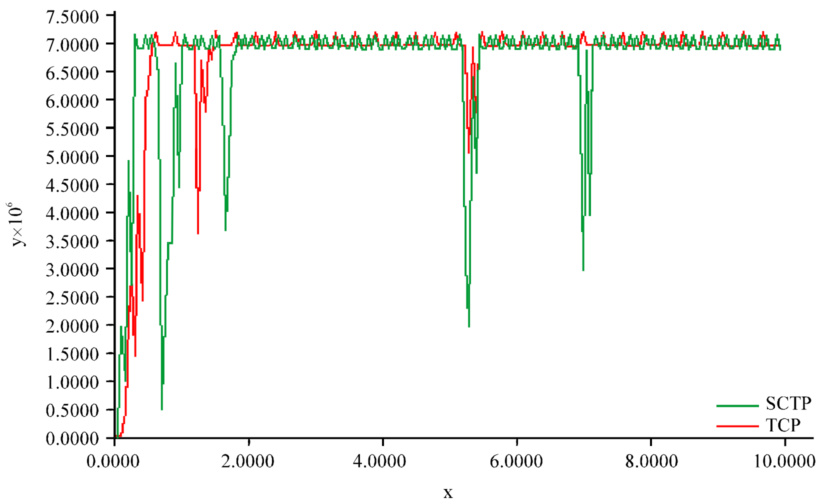


Fig. 24: Plotted throughputs of SCTP vs. TCP

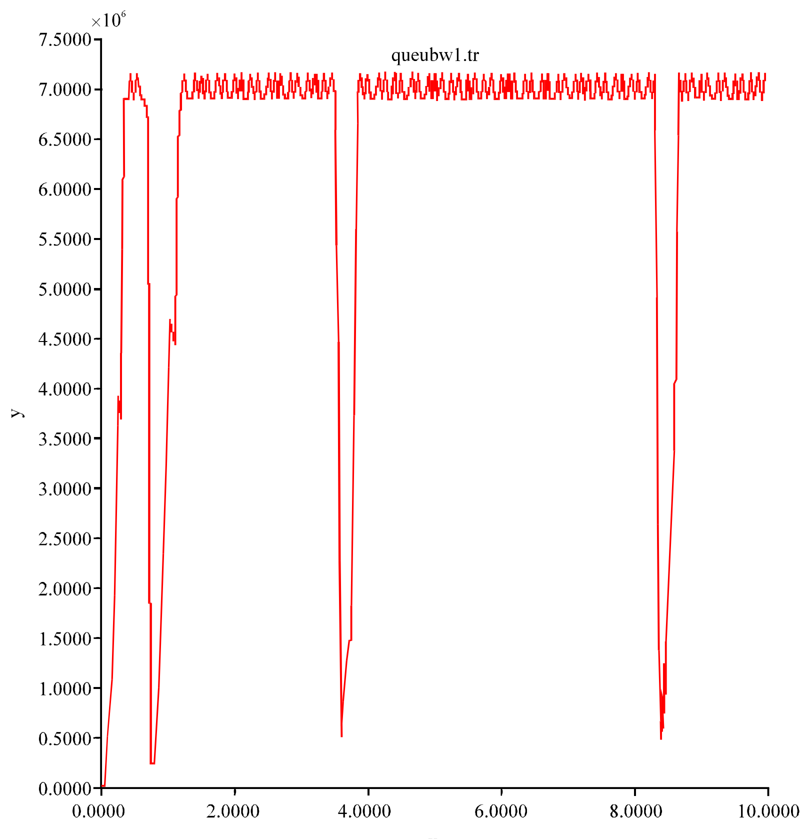


Fig. 25: Ideal TCP throughput

the output is almost ideal and the represented model near the high traffic inside the transport layer which experiences numerous congestion events.

Figure 25 and 26, respectively show the throughput of SCTP and TCP. Both figures differ from Fig. 24 which shows the first scenario of the workload.

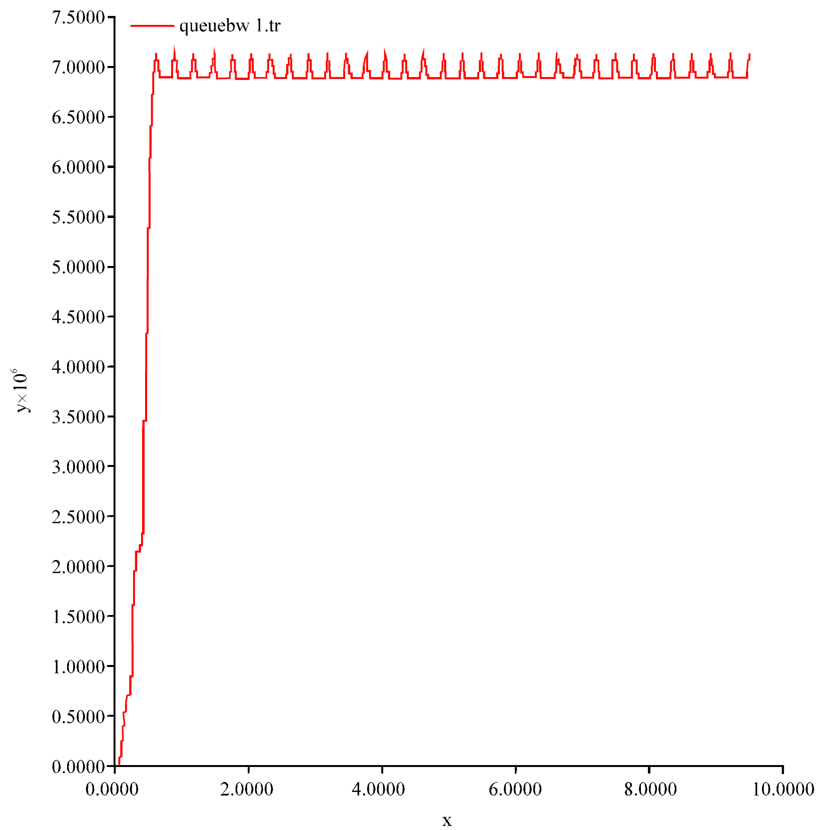


Fig. 26: Ideal SCTP throughput

The difference between performance factor measurements with regard to either the ideal state or the workload state distinguishes between performance factor behavior in the ideal environment or workload environment. When considering the differences, the ideal TCP throughput shown in Fig. 25 and the ideal SCTP throughput shown in Fig. 26 are compared with the workload throughput shown in Fig. 24.

The ideal TCP throughput dropped by three points, as shown in Fig. 25. The workload TCP dropped by four points, as shown in Fig. 24. Moreover, the time period of the congestion avoidance state in the workload environment is greater than that in the ideal state.

The ideal SCTP throughput illustrated no drops (Fig. 26). By contrast, in the workload environment, the SCTP experienced a drop by two points (Fig. 24).

We point out that the results shown in both Fig. 25 and 26 have the same topology in the first scenario but without workload pressure which occasionally experiences drops or congestion point level. Clarifying the factor related to performance besides cwnd would provide a closer look into the behavior of TCP and SCTP.

Irrelevant to this study is the content of the throughput addition to packet loss. Packet loss refers to the vagrant packets that have not reached the correct address. This event happens when one or more packets of data exchanged across a network fail to arrive at their target.

Figure 27 shows packet loss TCP (green) and SCTP (red). Note that in this figure, the lower values represent better results, unlike all other figures wherein higher values represent better performance.

The red line represents SCTP packet loss, whereas the green line represents TCP packet loss. The highest line indexing high packet loss means that the highest probability of packet loss occurs more in the TCP. Figure 27 shows that high packet loss is affected by throughput rate and vice versa. Low packet loss rate signifies a highly successful delivery of packets to the final destination for SCTP.

OPEN ISSUES AND CHALLENGES

User-side limitations: To obtain a preferable performance, the bandwidth delay product of the TCP connection

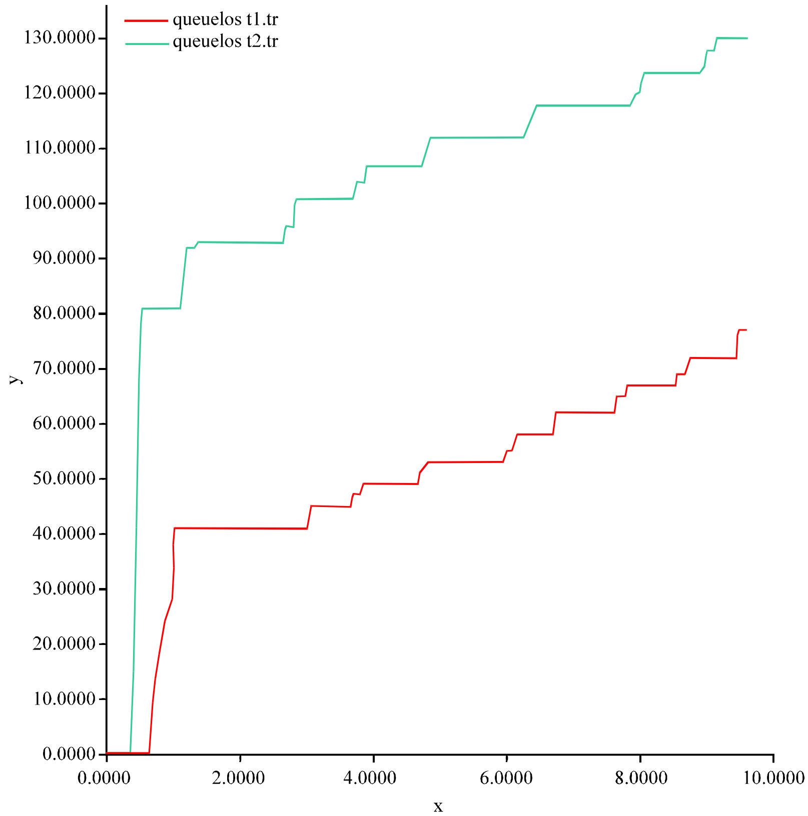


Fig. 27: Packet loss TCP vs. SCTP

should be larger than 65,535 bytes (Park *et al*., 2011). Therefore, the UE has to provide window scaling in the TCP.

Cost reduction: The TDD used both DL and UL single carriers in one stream simultaneously (Rumney, 2013; De La Roche *et al*., 2012). This technology is augmented inside LTE-A. Given its low cost, LTE-A is now extensively used in the industry field, with minimal attempt to operate at almost negligible levels (Susitaival *et al*., 2010).

LTE-A limitation: LTE-A deployed DL and UL through MIMO. TDD does not implement MIMO inside LTE-A (Li *et al*., 2010). Therefore, TDD needs to deploy MIMO inside LTE-A.

Filter selection challenges: LTE-A used MIMO at 2×2 for UL and 4×4 or 8×8 for DL. Hence, the next generation or enhanced LTE-A (Rumney, 2013) has to provide N×N that is, N = 10 MIMO. However, this requirement may be challenging because of the narrow space between duplex spacing and space gap, making it difficult to design a filter to prudently transmit the regrowth filter to detect DL signal or UL signal.

UDP-based implementation challenges: The transport layer included many protocols. Expanding new mobile technology, such as LTE-A, needs high-performance web traffic and high-resolution video streaming (Kishiyama *et al*., 2013). The implemented enhanced UDP over LTE-A will support new features of the high-resolution video streaming mechanism.

LTE-A carrier modulation challenges: The bearer service LTE-A carries out only packet switch and not circuit switch (Rumney, 2013); hence, the VoIP must be used for voice communication. This application requires priority handling or some level of QoS implementation.

CONCLUSION

We analyzed the performances of protocols of the transport layer, such as various TCP protocols (e.g., ordinary TCP, Reno, New Reno, Vegas and FACK) and the proposed protocol SCTP in the cwnd. of these protocols over the LTE-A network and wire networks. The results obtained from reviewed studies illustrate unequivocally that the TCP protocol can be considered a vital nerve in the transport layer. In spite of developments which include the creation of the congestion algorithm, TCP cannot keep up with the growing new high-performance technology. Therefore, SCTP will be an effective switch. This result agrees with that of other research.

We implemented two scenarios containing eight protocols working inside the transport layer. The first scenario worked over LTE-A with 10 UE nodes and the second scenario connected five nodes as senders through two bottleneck gateways to five servers. Figure 21 and 23 show the comparison of the cwnd of the SCTP protocol with that of other TCP protocols. Other performance factors, such as throughput and packet loss, were also measured.

SCTP provides important features, including multi-homing and multi-streaming. Multi-homing guarantees a seamless and redundant connectivity approach. Multi-streaming divides the physical channels into streams between sender and receiver.

The results highlight two important factors when passing packets successfully within a predetermined time period over well-known network parameters.

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