

PERFORMANCE ANALYSIS OF VoIP SERVICES IN LTE NETWORKS

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Abstract: *Future telecommunications nodes will be predominantly mobile, bringing a new development of multimedia capabilities and performance service areas compared with the communications carried on fixed networks. In this paper there will be analyzed the performance of VoIP services in LTE networks, depending on the bandwidth of the channel, the method of modulation and coding schemes, by simulation of two techniques for programming packages: FD (Fully Dynamic) and SPP (Semi Persistent Packet).*

Key words: *VoIP, LTE, Modulation Coding Scheme, Network simulation.*

1. Introduction

Development of Internet, the growing demands for integration of traffic video, audio and data telecommunications market have led to transformations that no other engineering area has ever known. Challenges arose when it became mandatory to find solutions to integrate technologies of the Internet, based on intelligent user terminals, technologies of traditional telecom (having "intelligence" as part of the network) with the security, safety and quality insurance functions tested and proven over time.

When VoIP technology will be fully introduced in mobile networks based on LTE, it is expected a quality of calls at least comparable or even better than circuit-switched based services.

This requires thorough analysis of the performance in LTE networks, in connection with sequencing techniques and programming packages for the future VoIP services.

IP-based services are already used in the

standard HSPA (High Speed Packet Access) of 3GPP, but the importance of IP services will be even greater for LTE networks that support only the PS (Packet Switching).

Some recent works have studied the performance of VoIP services in LTE networks, focusing on the programming packages semi-persistent effect on service quality VoIP [10] and on multi-user different programming strategies in the context of OFDMA technology such as: FS (Fair Scheduling) [9], DSA (Dynamic Subcarrier Assignment) or APA (Adaptive Power Allocation) algorithm [4].

2. Packet Programming Techniques

LTE works on packet switching, organized at 2nd OSI Layer using FD programming that allocates available resources of the user equipment separately for each transmitted packet. Transmission resources are composed of PRB (Physical Resource Blocks) and MCS (Modulation Coding Scheme) values.

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The selection depends on the number of PRB channel bandwidth and MCS values that determine the transmission speed and capacity of the PRB community. PRB allocation of sites and selection of the MCS's mobile terminals are indicated by the PDCCH (Physical Downlink Control Channel). LTE uses a 1 ms interval as the minimum time unit assigned - the TTI (Transmission Time Interval), where a TTI consists of two time-slot every 0.5 ms.

Explanation of used terms

PRB. Frequency domain structure of LTE is based on resource blocks consisting of 12 subcarriers with a total bandwidth of $12 \times 15 \text{ kHz} = 180 \text{ kHz}$. Specific to the radio access in LTE, the transmission path configurations allow for 6 to 110 RF resource blocks on a carrier. This allows channel bandwidths from 1.4 MHz to 20 MHz in steps of 180 kHz, thus ensuring the spectral flexibility [6]. His bandwidths of the channels are presented in Table 1.

Channel characteristics Table 1

Channel bandwidth [MHz]	The number of resource blocks
1.4	6
3	15
5	25
10	50
15	75
20	100

MCS. Its order numbers (e.g. MCS = 9 or MCS = 15) were used to extract the bit rate of the wireless connections from the MCS index [13].

MCS values summarize the number of spatial streams transmitted, the type of modulation and coding rate.

3. Blocks Transportation

Considering the uplink direction, data is first received by the PDCP (Packet Data Compression Protocol) layer, which

transmits data compression at the RLC (Radio Link Control) next level. It will merge or segment data into blocks of suitable sizes and submit them with their own MAC header. This level will select the schema modulation and coding set at the physical level (Figure 1).

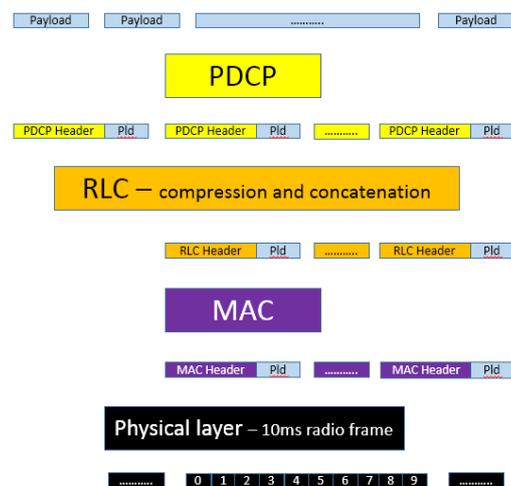


Fig. 1. Transport blocks

The number of bits transferred in a block per millisecond depends on the PRB and on the MCS, both of which are assigned to the mobile terminal device (UE - User Equipment).

3.1. Code Rate

Code rate may be defined by efficiency of data transmission or, in other words, the ratio between the transmitted bits and the maximum amount of bits that can be transmitted in a transport block.

Code Rate = $(\text{TBS} + \text{CRC}) / (\text{RE} \times \text{Bits per RE})$,

where: TBS = Transport block size; CRC = Cyclic Redundancy Check bytes added for error detection; RE = Resource elements for respective radio channel; Bits per RE (according to the MCS).

3.2. HARQ Processes

HARQ - hybrid ARQ (automatic repeat request) uses a "stop and wait" protocol. When a transmission is made, the entity that issues it stops and waits until it receives an acknowledgment (ACK) or denial (NACK) from the receiver, and then transmits the next block of data or retransmits the same block. In either case (ACK or NACK), the broadcaster must schedule and process the next transmission in a certain time-frame. If LTE FDD (frequency-division duplex) is used in the uplink, this interval is set at 8 sub-frames each 1ms. Because it is necessary that only a sub-frame should transmit data, there are seven frames that do not use sub-band transmission. For fully utilization of this band, LTE uses multiple parallel HARQ processes, de-phased from each other. Each process transmits a data block. By the time the next allocation of transmission is received, the ACK or NACK from the reception will already have been received and the next package for (re-) transmission would have been created (Figure 2).

LTE uses asynchronous transmission downlink HARQ Type-II. This means that the receiving party does not in advance what (or when) it is transmitted, so that a HARQ process identifier and RV (Redundancy Version) must be sent along with the data. RV shows what kind of combination of data and bit error correction (FEC - Forward Error Correction and ED - Error Detecting) is transmitted to the mobile terminal. This is done using resource allocation messages

specific for the PDSCH channel (Physical Downlink Shared Channel), sent using the channel PDCCH (Physical Downlink Control Channel).

The advantage of this scheme is that the scheduling algorithm has more freedom in deciding to which mobile devices data will be sent using any sub-frames.

Unlike the downlink, the LTE uplink transmission is using a synchronous HARQ. Here, the eNB (eNodeB - evolved Node B) knows exactly which HARQ process and which RV will be transmitted to the mobile terminal in the future. Such information should not be included in the PDCCH message radio channel as planning information about uplink to the mobile terminal is not particularly needed.

Synchronism is possible because the mobile terminal transmits the same HARQ process every 8th sub-frame.

4. Adaptive Modulation and Coding

Adaptive modulation and coding is trying to adapt the process of HARQ transmission to channel conditions. Under strong signal, less redundancy bits are used and/or a higher order modulation is in the initial transmission, allowing a greater flow of information in a given band. In weak signal conditions, more bits of redundancy are used and/or a lower order modulation, in order to increase the likelihood of correct reception.

This, however, decreases the flow of information to the user. If the error rate is 0, then it can be concluded that too much protection is used. Alternatively, if sufficient

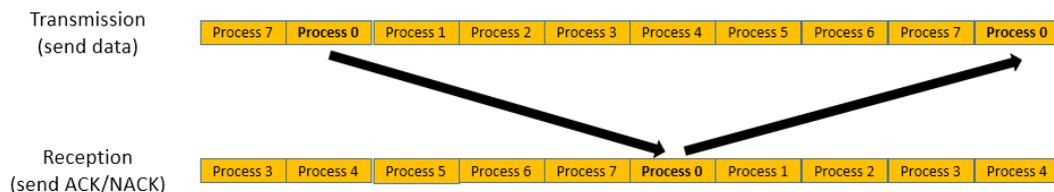


Fig. 2. *Transmission flow*

protection is not used, the same data will be retransmitted, wasting unnecessary network resources. The ideal situation is where the throughput is maximized under conditions of relatively low error rates (small but greater than 0).

Like the W-CDMA technology, LTE eNodeB sites decide what type of modulation will be used depending on what kind of information is transmitted to the mobile terminal by the LTE CQI - Channel Quality Indicator.

Full Dynamic Scheduling (FD) is very flexible and allows the processing of bursts of random data, offering a consumption limited by bandwidth (for applications such as e-mail, video streams or web browsing) but is less suitable for applications using real time transmission flows, such as voice.

To reduce the signaling control for VoIP traffic, it was proposed the so-called Semi-Persistent Scheduling (SMP) for initial transmissions and dynamic scheduling for retransmission.

The semi-persistent allocation technique is required until the user equipment receives another indication from the control channel, which can occur when channel conditions have changed. Allocations for initial transmissions are supported by PDCCH and PUCCH (Physical Downlink Control and Uplink Channels).

Retransmissions are programmed using dynamic control channels on L1 and L2 levels.

The simulation uses 3 PRBs for HARQ retransmissions using periodic allocation of 5 sub-frames for full dynamic programming.

5. VoIP Services Model

Using the VoIP protocol to transfer voice packets over the network, for the first session QoS is negotiated between the pair of entities that are using SIP (Session Initiation Protocol) [11] and SDP (Session Description Protocol) [7].

During VoIP transfer a user session can be in an active state (talk time) or inactive (silent period). The duration of each state is exponentially distributed in bursts of 0.65 s and 0.352 s respectively.

Using a combination of OPNET simulation and MATLAB post-processing, we have chosen G.711 (64 kbps) and G.723.1 (12.2 kbps) vocoders (voice coders). The size of the package payload was 160 bytes for G.711 and 40 bytes for G.723.1. The generation time intervals were 20 ms and 30 ms respectively.

The packet arrival SIDI (Silence Insertion Descriptor Interval) was 160 ms for both codecs. For L2 (Level 2 OSI) header 6 bytes were added for each packet and 2 bytes for the RTP/UDP/IP compressed header.

Voice Activity Detection (VAD), Discontinuous Transmission and Comfort Noise Generation (CNG) were also applied.

System capacity was defined as the maximum number of users in the cell, the average packet delay was 100 ms and the mean saturation buffer was less than 2%.

5.1. Simulation Model Parameters

The network model included an eNB, an ePC (evolved Packet Core), and a communication server connected via an IP link speed of 1 Gbps. The service users were randomly positioned within a radius of 1 km. The radio channel between each user and eNB was calculated based on the M.1225 path attenuation model recommended by ITU-R, for indoor and outdoor walking test conditions. It was considered acceptable a fading attenuation of 10 dB for outdoor and of 12 dB for indoor. The antenna transmission reception gain was estimated at 10 dBi for outdoor and walking conditions and at 2 dBi for indoor conditions. The noise reception was considered to have the value of 5 dB and the thermic noise

density to have the value -174 dBm heat per Hz. The attenuation penetration through buildings was 12 dB with a standard deviation of 8 dB. Other attenuation (in cables or connectors) was estimated to 2 dB.

For the initial access-, originating-, terminating-, network registration-procedures it is used a so-called Random Access Channel (RACH). The user is using this procedure to send the control messages over the Common Control Channel (CCCH) to the eNB during the Radio Resource Control (RRC) connection setup, in order to connect to the network. If the user has no time-slot assigned on PUCCH for programming transmission requests, the same procedure is used for transmission of BSRs (Buffer Status Reports) towards the eNB. There are two types of random access procedures: RACH based on conflict and RACH without conflict, in terms of packet collisions. In the simulation performed it was used RACH without conflict because there it was no possibility for collisions in the preamble.

This procedure has 3 steps:

1. The assigning of the random access preamble: eNB assigns the 6 bit code as a preamble
2. The sending of the random access: the user transmits the assigned preamble.
3. The random access response.

In the simulation there were used a total number of 64 random access preambles. The BSR is required to inform the network about the throughput on the UL (Uplink). BSR will generate a MAC control element containing information about the volume of data available for transmission in the RLC and PDCP level (Packet Data Convergence Protocol). BSR will be started when the UL data becomes available for transmission, belonging to a logical channel with a higher priority than the data already

existing in the user's transmission buffer. In this case, the BSR is called BSR normally set. When BSR is ON but no resources are allocated for a new transmission, the user will then start the procedure SR (Scheduling Request) to request UL resources; for example, an UL-SCH (Uplink Shared Channel) for transmitting the MAC control element. Only BSR normally adjusted can start the SR procedure when the user does not have allocated resources for a new transmission in the current TTI interval. If the network is not able to receive the BSR MAC control element when it is submitted, there will be no priority for data transmission (e.g. when the user does not have enough resources on UL and enters the "deadlock" state). In this case it will be applied a BSR retransmission procedure that uses a timer to increase transmission reliability of the BSR. The user starts the timer when the control element MAC of the BSR is received and restarts it when resources are allocated to UL for a new transmission.

When the timer expires and the user has available data for the buffer, the user equipment will start a normal BSR.

In the simulation it was used a Round Robin (RR) packet scheduler in the time domain and a proportionally balanced stream packet scheduler in the frequency domain. The simulation parameters are listed in Table 2.

The eNB allocates physical layer resources for the shared channels UL-SCH and DL-SCH. The resources are composed of PRB and MCS, where MCS determines the bit rate and capacity of each PRB.

To analyze the performance of VoIP services in LTE networks a number of scenarios were performed [12].

These were, according with Table 3 in which we introduced different values of the parameters listed in Table 2.

Parameters used in simulations

Table 2

Parameters		Value
PHY Profile	Operation Mode	FDD
	Cyclic Prefix	Normally 7 symbols/timeslot
	EPC	348 kbps
	Carrier Frequency	2GHz
	Subcarrier in-between space	15kHz
BSR Parameters	Periodical Timer	5 subframes
	Retransmission Timer	2560 subframes
L1/L2 Parameters	Reserved dimension	2 RBs
	Allocation Periodicity	5 subframes
HARQ Parameters	Maximum Numbers of Retransmissions	3 (UL&DL)
	Retransmission Timer HARQ	4 subframes (UL&DL)
	Maximum Numbers of HARQ Processes	8 (UL&DL)

Simulation Scenarios

Table 3

No.	Simulation description		
	Channel	Link adaptation	Packet programming
1	5 MHz ideal	MCS 9	FD
2	5 MHz ideal	MCS 9	SMP
3	5 MHz ideal	MCS 15	FD
4	10 MHz ideal	MCS 9	FD
5	5 MHz real	Link adaptation	FD
6	5 MHz real	Link adaptation	FD

For the simulations of an ideal channel, neither channel adjustments nor HARQ have been used (not being necessary in the absence of errors).

There were used two voice codecs, G.723.1 and G.711 and semi-persistent programming to observe their influence over VoIP capacity.

For a real transmission, channel adaptation was used. The signal to interference noise ratio (SINR) was calculated for each user.

The link adaptive algorithm seeks to maximize the spectral efficiency by choosing the best MCS for a given SNR (the threshold of -10 dB Bit Error Rate - BER given by the Additive White Gaussian Noise - AWGN - curves for each MCS), while the LTE standard defines 29 values for the MCS.

However the user has only a 4 bits response to indicate the preferred MCS, so

the eNB can only use the first 15 levels from the MCS list.

In the simulations that are not using HARQ the forward error correction (FEC) was not used, the packets affected by errors being retransmitted (up to 3 retransmissions), while in the simulations with HARQ eight “stop-and-wait” processes were used together (by the combination between FEC and ARQ), both on the uplink and the downlink [1-2].

When the received data can not be perfectly decoded, the receiver requests retransmission.

6. Simulation Results

- *The LTE network capacity for VoIP services under ideal conditions*

In the absence of errors in the channel, and when BER and PER (Packet Error

Rate) are considered invalid, the graph in Figure 3 shows the LTE network packet delay for $B = 5$ MHz with the link adaptation value $MCS = 9$, for the codec G.711 and G.723.1.

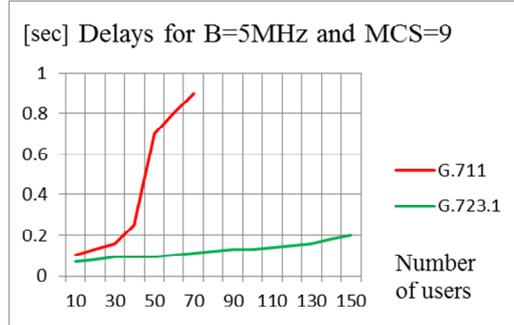


Fig. 3. Packet delay for an ideal network with $B = 5$ MHz and $MCS = 9$

The Figures 4 and 5 show the packet loss as a percentage based on the number of VoIP users, both for uplink and downlink. It is observed that the downlink obtains a higher capacity network service for VoIP users, for a smaller packet loss [5].

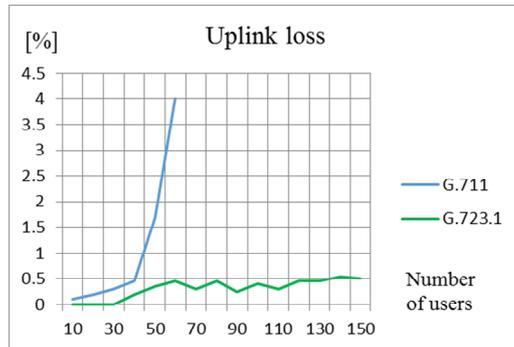


Fig. 4. Average packet loss for UL

The VoIP quality of connections in LTE networks can be assessed by measuring the head-to-head delays and the percentage of packet loss. Using the G.723.1 codec the capacity can increase up to 150 users per cell, in terms of delays up to 200 ms and with a packet loss rate up to 0.5%. When small delays are taken into consideration the number of users per cell is reduced.

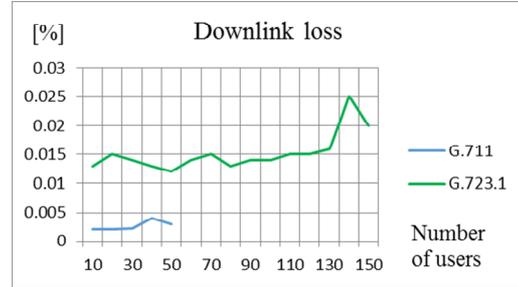


Fig. 5. Average packet loss for DL

The absolute capacity of a link from the VOIP users point of view can be determined when absolute values of QoS parameters are imposed. The network performance in terms of packet loss and delays is influenced by the bandwidth used and by the MCS values, as shown in Figure 6, where two bandwidths of 5 MHz and 10 MHz have been used, and for MCS values 9 and 15. It is observed that using a higher bandwidth (10 MHz) leads to a smaller packet delay (Figure 7). The 5 MHz link presents a longer delay for both values of MCS, which corresponds to the theoretical channel capacity calculated using the modified Shannon formula:

$$R = B\eta\log_2\left(1 + \frac{SNR}{\eta SE}\right),$$

where B = bandwidth;
 SNR = signal to noise ratio;
 ηBW = bandwidth efficiency;
 ηSE = spectral efficiency.

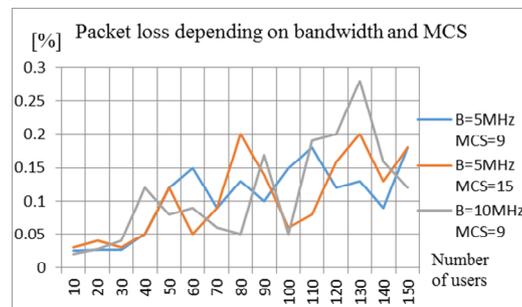


Fig. 6. The total packet loss depending on bandwidth and MCS

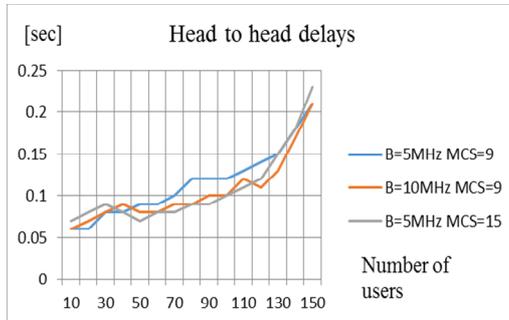


Fig. 7. Head-to-head delays depending on bandwidth and MCS values

LTE bandwidth efficiency is reduced because of the following reasons:

- ACL - Adjacent Channel Leakage;
- Reference signals;
- Synchronization signals;
- RAP (Random Access Preamble);
- Control packets headers from layers L1 and L2.

In simulations it was considered that the bandwidth effectiveness is 0.9 due to ACLR (ACL Ratio) value and the effectiveness of the channel downlink is 0.62 and of the uplink is 0.78 due to headers involved in the control channels [10]. Spectral efficiency was chosen for both downlink and uplink to be $\eta_{SE} = 0.88$ [3]. In terms of QoS parameters, the packet head-to-head delay was chosen to be less than 100 ms and the mean overflow of the reception buffer to be less than 2%.

Thus for a LTE network with $B = 5$ MHz and $MCS = 9$, the number of users of VoIP services is 70, for $B = 5$ MHz and $MCS = 15$, network capacity increases to 100 users, and for $B = 10$ MHz and $MCS = 9$ the number of users is 115, which corresponds to the theoretical value of the channel capacity obtained with the modified Shannon formula. Figure 7 shows the behavior of the average total packet loss and average end-to-end delay for the LTE network after the application of SMP and FD techniques.

There is an increased user capacity of VoIP services when using SMP. It can be noticed that when using SMP for a 5 MHz bandwidth, the performance is equally effective on the number of users, as in the case of using FD for 10 MHz bandwidth [8]. Using SMP also results in an increase of packet loss, but decreases their delay. In terms of packet loss, the percentage is similar in SMP or FD enforcement cases.

• LTE network capability for VoIP services in real conditions

The study of VoIP service capabilities under real conditions is based on the impact of the number of users on the main QoS parameters (packet loss and delays).

Compared to ideal conditions, real packet loss is high, due to the absence of HARQ, and delays can rise to unacceptable values for real-time services because of links' non-adaptation.

For a variable number of users, Figure 8 shows the percentage of total losses behavior for three types of simulations: no HARQ and MCS, without HARQ but with MCS and with HARQ and MCS.

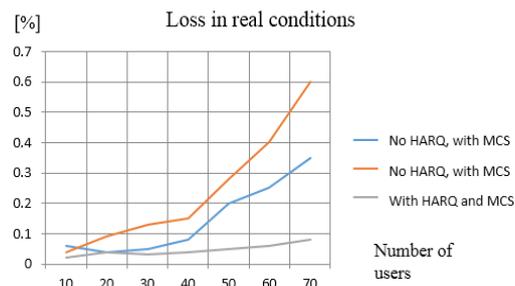


Fig. 8. Total loss by using or not using HARQ and MCS

One can observe the effect of applying HARQ and MCS by obtaining a small percentage of losses even at a relatively large number of users.

The study of the application of HARQ and MCS delays in terms of voice packets is shown in Figure 9.

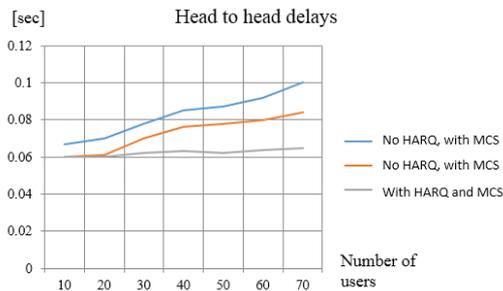


Fig. 9. Total delays by using or not using HARQ and MCS

Compared to the ideal case, when the absence of errors was presumed, in reality the HARQ FEC effect can be observed, where network delays and losses are decreasing through error detection and lower percentage of packet loss.

7. Conclusions

Performance analysis of voice services based on packet switching in an LTE network, was carried out through simulations in ideal conditions and real channel transmissions.

Network performance was highlighted by using two technical models, SMP and FD which offer lower or greater advantages respectively, in terms of QoS.

When using SMP, network capacity is not limited due to control channel resources, but it is limited by bandwidth. From this point of view, the use of SMP may be recommended for delay-sensitive applications (VoIP), but it is not recommended for data transmission applications (web or video) that are bandwidth consuming.

Also, contrary to assumptions that the use of ARQ packet switched networks for real-time services would not be generally satisfactory, it was demonstrated that the introduction of HARQ technique has brought benefits in terms of QoS.

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