Bulletin of the *Transilvania* University of Braşov • Vol. 12 (61) No. 1 - 2019
Series I: Engineering Sciences
https://doi.org/10.31926/but.ens.2019.12.61.5

QoS STUDY IN AN USRP PRIVATE LTE NETWORK

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Abstract: *Long Term Evolution - LTE has been the dominant mobile technology nowadays with the main advantages of high data rates, antenna diversity and spectrum flexibility. Efforts were made to emulate the LTE technology using SDR equipment, the most notable being OpenLTE, which is an open source implementation and which works using B200 USRP boards. This paper studies the quality of service parameters in a private LTE network which runs over multiple USRPs that act as eNodeBs and OpenLTE provides the core part of the network.*

Key words: *USRP, LTE, QoS, MOS value, QoE.*

1. Introduction

3GPP Release 13 [10] sets as its latest standard the 3GPP Long Term Evolution (LTE) regarding the mobile communications. Unlike UMTS, it does not use WCDMA as a transmission scheme, but uses OFDMA (Orthogonal Frequency Division Multiple Access) and SC-FDMA (Single Carrier FDMA) for uplink and downlink traffic, respectively. Among the most important features of these releases there are enhanced small cells (higher order modulation, dual connectivity, cell discovery, self configuration), carrier aggregation (2 uplink carriers, 3 downlink carriers, FDD/TDD carrier aggregation), MIMO (3D channel modeling, elevation beamforming, massive MIMO), New and Enhanced Services (cost and range of MTC, D2D communication, eMBMS enhancements).

LTE's data rates are up to 100 Mbits/s for downlink and up to 50 Mbits/s for uplink, and the latencies in the radio access network are 10 ms. LTE was designed around the intent of having spectrum flexibility, having in mind bandwidths ranging from 1.4 MHz to 20 MHz and many usable frequency bands including the 700 MHz band which permits better indoor usage. LTE is not compatible with the old 2 and 3G technologies, GSM and UMTS, but it is aimed to meet the requirements of the 4G radio access network of 1 Gbits/s for static applications and 100 Mbits for mobile applications.

There are no more different nodes as there were in GSM and UMTS, the eNodeB (evolved NodeB) is the only node type. It deals with all the radio interface-related functions. Regarding the EPC (Evolved Packet Core), the other entities are the MME (Mobile Management Entity), which deals with mobility and security functions, and with UE (User Equipment) identity, the SGW (Serving Gateway) which wraps up the interface

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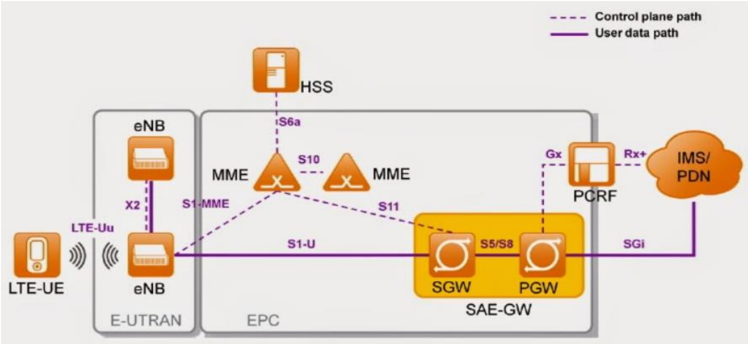

Fig. 1. *The basic architecture of a LTE network* [14]

towards E-UTRAN (Evolved UTRAN), and the PGW (Packet Data Network Gateway) which wraps up the interface towards the PDN.

LTE has no voice support. There are 4 solutions for providing voice over LTE, namely the use of third parties voice services such as Skype [13], CS (circuit switched) fallback [8], VoLGA (Voice over LTE Generic Access) [12], which can be roughly described as CS over PS, SRVCC (Single Radio Voice Call Continuity).

There is also the VoLTE (Voice over LTE) initiative [11], which revolves around the idea of using IMS (IP Multimedia Subsystem) for voice support in LTE.

This paper attempts to provide a study in terms of MOS (Mean Opinion Score) values regarding different bandwidths and codec types, all these viewed from the perspective of the user which employs Mobile VoIP applications.

2. Previous Studies

In LTE networks, standard solutions for voice and SMS services were analyzed by Gavrilovic [2]. Qualitative comparison followed between IMS Telephony (MMTel), IMS Telephony with handover to CS domain (SRVCC), and CS fallback. From the author's words, although SRVCC manages the problem of initial spotted LTE coverage well, MMTel creates possibilities for new service offerings like video calls, chat, and image sharing. Lastly, Gavrilovic has come to the conclusion that the LTE rollout from IMS/MMTel rollout can be decoupled by operators due to CS fallback features.

Numerous methods for delivering VoIP services in LTE networks including IMS Telephony with handover to CS domain, CS Fallback and Voice over LTE via Generic Access (VoLGA) were quality-wise compared by Paisal [5]. They decided that for operators that do not wish to deploy IMS, CS Fallback and/or VoLGA would be of most interest.

Single Radio Voice Call Continuity (SRVCC) provides an interim solution for handing over VoLTE (Voice over LTE) to 2G/3G networks. The voice calls on LTE network are meant to be packet switched calls which use IMS system to be made.

For voice calls based on IMS and VoLGA, the solution is to use SRVCC when the subscriber switches to the 2G or 3G network. From the CS fallback point, the voice call will be interrupted, so the call will be initiated on the 2G or 3G network.

The mobile network operators owning 2G/3G networks and planning to deploy IMS are contemplating about CS fallback or VoLGA as a transitional step, but later when IMS has been installed, they will use SRVCC. Paisal consider that mobile network operators deploying IMS together with LTE should reach SRVCC straight.

VoIP performance in LTE downlink (DL) using the Adaptive Multi-Rate (AMR) [9] 12.2 codec was significantly studied in four simulation cases standardized by 3GPP, according to Henttonen et al. [3]. The effects of control channel capacity, packet bundling, link adaptation and number of Hybrid ARQ (HARQ) processes on VoIP capacity were analyzed particularly. In an LTE network, it was also believed that link adaptation along with packet bundling increase the overall VoIP capacity. In additional to this, in this paper has been demonstrated that control channel limitations can be rewarded by packet bundling.

3. Groundwork for the Experiment

I_s symbolizes impairments to the source signal, R_o is the basic signal-to-noise ratio defined as 93.2 [7], and I_d is the impairment due to delay and echo effects. $I_{e\text{-}eff}$ represents impairments due to packet losses.

Moreover, *BurstR* is the mean number of packets lost in a burst of lost packets at each second, while *Ppl* symbolizes instantaneous packet loss rate. *Bpl* is the strength factor to random packet loss and was set to *Bpl* = 20 for the GSM Fullrate (FR) codec [12].

The experimental parameters can be found in Table 1. Interpretation of MOS values can be seen in Table 3.

Simulation setup Table 1

Parameter Description	Parameter Values
System bandwidth	{5,15} MHz
VoIP codecs	{G.711, G.723.1, G.729 A}
Duplex mode	FDD
Base frequency (UL)	1920 MHz
Base frequency (DL)	2170 MHz

MOS values results after simulation Table 2

VoIP Codec	System Bandwidth	
	5 MHz	15 MHz
G.729 A	3.12	3.13
G.723.1	2.76	2.77
G.711	3.75	3.74

Meaning of MOS values Table 3

MOS Value	Detected problems
5	inconsiderable
4	considerable, not disturbing
3	a little disturbing
2	disturbing
1	very disturbing

In LTE networks, the effects of using robust header compression (ROHC) procedures were investigated by Puttonen et al. [4] in huge mobile scenarios for VoIP applications. Unlike near-stationary (3 km/h) scenarios, the capacity loss from highly mobile (120 km/h) scenarios was reported to reach 65%.The overall capacity loss of using non-ideal ROHC related to using ideal ROHC reached 1-7% was also reported by the authors.

Andersson and Åhlund [1] proposed a prototype elevating multimedia quality of experience (QoE) using statistical and prediction learning. In conclusion, it was determined and evaluated an access network selection system for mobile nodes in a combined WLAN/LTE environment. The main optimized parameter in terms of user-perceived quality of service was VoIP performance.

4. USRP Setup

The scope of the experiment is to study the quality of service of VoIP applications in a private LTE network. The private network was accomplished by two Ettus B200 USRP which covers RF frequencies from 70 MHz to 6 GHz, has a Spartan6 FPGA, and USB 3.0 connectivity. This platform enables experimentation with a wide range of signals including FM and TV broadcast, cellular, Wi-Fi, and more. The USRP B200 features one receive and one transmit channel in a bus-powered design. Because the B200 is enabled with USRP Hardware DriverTM (UHD), users can develop their applications and seamlessly port their designs to high-performance or embedded USRPs UHD is an open-source, cross-platform driver that can run on Windows, Linux and MacOS. It provides a common API, which is used by several software frameworks, such as GNU Radio. The application that was used, OpenLTE, is an open source implementation of the 3GPP LTE specifications. It is installed and it's running under a Linux SUSE 15 operating system.

12 SIMs were used along with 12 Samsung phones. Chosen bandwidths were of 5 and 10 MHz. The USRP settings and parameters can be found in Figure 2.

As can be seen in the lower part of the picture, users are added using the IMSI number, the IMEI number and the private key K.

The USRP coverage is equivalent to the area of two LTE cells. The phones are initiating VoIP calls between themselves and between cells so at all times there are 5-6 calls going on at the same time.

MOS values were used as the studied metric.

Using the E-model [6] such values can be calculated according to the following formulae:

$$R = R_0 - I_S - I_d - I_{e-eff}, \tag{1}$$

$$I_{e-eff} = I_e + (95 - I_e) \times \frac{P_{pl}}{P_{pl}/BurstR + B_{pl}}, \tag{2}$$

$$For\, R < 0: MOS = 1, \tag{3}$$

$$For\, R \in [0,100]: MOS = 1 + 0.035 \times R + R \times (R - 60) \times (100 - R) \times 7 \times 10^{-6}, \tag{4}$$

$$For\ R > 100: MOS = 4.5. \tag{5}$$

```
        System Parameters:
                band = 1
                bandwidth = 10
                cell_id = 1
                debug_level = radio phy mac rlc pdcp rrc mme gw user rb timer iface msgq
                debug_type = error warning info debug
                dl_center_freq = 2112500000
                dl_earfcn = 25
                dns_addr = C0A80101
                enable_pcap = 0
                ip_addr_start = C0A80102
                mac_direct_to_ue = 0
                mcc = 001
                mnc = 02
                n_ant = 1
                n_id_cell = 0
                p0_nominal_pucch = -96
                p0_nominal_pusch = -70
                phy_direct_to_ue = 0
                q_hyst = 0
                q_rx_lev_min = -140
                rx_gain = 35
                search_win_size = 0
                sib3_present = 0
                sib4_present = 0
                sib5_present = 0
                sib6_present = 0
                sib7_present = 0
                sib8_present = 0
                tracking_area_code = 1
                tx_gain = 100
                ul_center_freq = 1922500000
                ul_earfcn = 18025
                use_cnfg_file = 1
                use_user_file = 1
start
ok
print_users
ok 2
imsi=226031405080987 imei=352937081971696 k=05298E362D8C2CA39D11CD312B599E13
imsi=226031405080995 imei=354735076444093 k=1D7F54DC401A8AB41718824F95A75B71
```

Fig. 2. *USRP setup parameters*

5. Results

The USRP private LTE network traffic generated different values according to the codec used. The final results can be found in Table 2. Clearly, higher MOS values will be obtained using high bitrate bit codecs (the G.711 with a nominal bitrate of 64 kb/s). In the same direction, lower MOS values will be obtained using more low-bitrate codecs (G.723.1 5.3K, and G.729A).

6. Conclusions

In the present the most important service brought in commercial LTE networks is mobile broadband. Voice services in LTE are still being defined and standardized. Standardization of voice services in LTE is a very important concern because voice is a very valuable feature in all telecommunication networks and is also the main income source for mobile network operators, at least until 5G testing is finished.

Our simulations demonstrate that a diversity of codecs may be used for delivering voice services with acceptable quality in LTE network.

We intend to do live experiments with a few commercial LTE networks to analyze how our simulated results correlate with experimental results. Besides, it is planned to extend this work and perform a lengthened study by adding different mobility scenarios to the simulation models, and, to also take the antenna diversity into consideration. The mobility scenarios would represent subscribers that are moving at walking speed or are in different means of transport, but that would prove difficult given the nature of the USRP private network which has to be extended.

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