Voice over LTE Performance Evaluation

Ahmed Amou El Arby and Ousmane Thiare

University Gaston Berger
Saint Louis, Senegal
Email: arby.amou@gmail.com
Email: othiare@ugb.edu.sn

Abstract. The 4G technology, known as Long Term Evolution (LTE), has been dominating the telecommunication market the last years offering a high data rate (up to 100Mbps) that will perfectly satisfy the users multimedia applications needs in term of bandwidth. Designed to be a fully packet switched network, LTE lack in offering the voice CS service. This paper propose an analytical model for simulating the performance of Voice over LTE (VoLTE) while presenting an overview of the latest up to date solutions for the voice over LTE such as Circuit Switched Fall Back, Voice over LTE via Generic Access and third party services for providing call continuity over LTE.

Keywords: LTE, SRVCC, VoLGA, CSFB

1 Introduction

During the past years, voice service was the primary revenue generator for cellular operators, though any migration in the network must support the transmission of the main service. Legacy network such as 2G/3G was basically relying on the Circuit Switched (CS) domain to offer the voice service and on the Packet Switched (PS) for transmitting data. Contrary to the previous cellular generations, LTE was designed to be All-IP network offering only the services based on the packet switching domain leaving the operators with a gap when it comes to serve the users who still have the 2G/3G phones.

Many solutions have been proposed to support the voice CS over LTE to ensure the service continuity. However, only one of them has been selected as a long term solution by 3GPP. This paper start by presenting an overview of the various up to date solutions in the telecommunication market that permit the voice over LTE such as the Circuit Switched Fallback (CSFB) [1]-[2], Voice over LTE via Generic Access (VoLGA) [3], Third Party Voice Over IP[6] and the Single Radio Voice Call Continuity (SRVCC) [4],[5]. Then, the paper also focuses on the long term solution SRVCC and proposes an analytical model for simulating the performance of SRVCC.

The rest of the paper is structured as follow: In Section II, we review the short-term solutions in LTE. Section III presents the SRVCC handover. In section IV, we

K.J. Kim et al. (eds.), *Mobile and Wireless Technologies 2016*,
Lecture Notes in Electrical Engineering 391, DOI 10.1007/978-981-10-1409-3_3

propose an analytical model that represents the handover delay interruption in SRVCC. Finally, we conclude the paper in Section VI.

2 Voice Over LTE short term solutions

2.1 Circuit Switched FallBack (CSFB)

The Circuit Switch Fallback is a short term solution that relies on the existing 2G/3G network. When a UE, camped on LTE, wants to make a CS call, it activate an Extended Service Request (ESR) to the MME. After receiving the ESR, the MME order the eNodeB to redirect the call to the legacy network 2G/3G. The UE receives the LTE Release Connection and camp on the legacy 2G/3G network and establishes the standard 2G/3G RRC Connection. The downside of this solution is the introduction of the delay due to the procedure execution. This delay varies based on the mobile originating the call, in this case the delay is around 4.7s [1], or terminating the call, in this case the delay is around 2.84s [1].

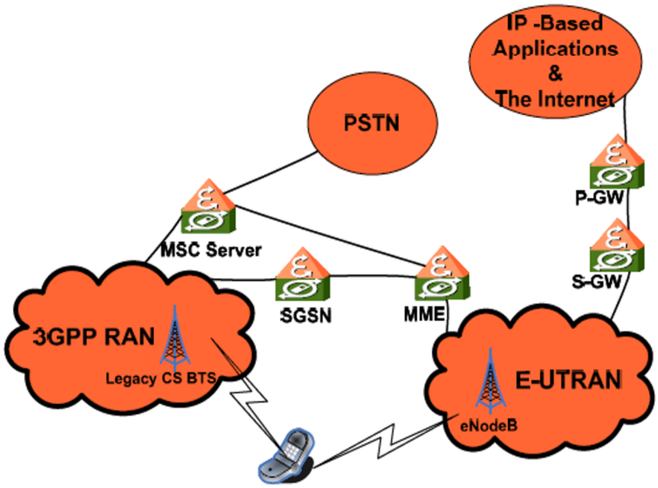

Fig. 1. CSFB network Architecture

Furthermore, the CSFB requires an introduction to new interfaces between LTE and legacy 2G/3G particularly the MME-MSC/SGSN and S-GW-SGSN [1] in order to perform the combined registration and paging process.

The benefit of using the CSFB is that the operators can use their existing 2/3G infrastructure to provide voice call.

2.2 The Voice Over LTE via Generic Acess (VoLGA)

The Voice over LTE via Generic Access (VoLGA) consists of having a second interfaces for Wi-Fi on the UE becoming dual mode devices that can connect to Wi-Fi over a 2/3G network when it is available and connects to the network operator through a GAN gateway. Similarly, the VoLGA use the LTE as a replacement of the Wi-Fi and introduce an entity named Access Network Controller (VANC) which is a mix between an IP based node and a base station controller node from the LTE and GSM/UMTS point of view respectively. The VANC connect to the 2G, 3Gand 4G via the same interfaces as per the figure 2.

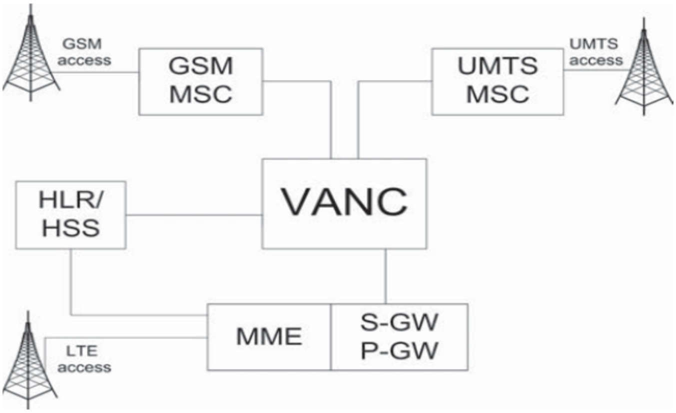

Fig. 2. VoLGA Network Architecture

The mobile start by registering to the LTE and then establishes a secure IPSec tunnel connection to the VANC over LTE [3]. Then, it registers to the 2G/3G through the VANC secure connection that has just been established. The translation between the PS LTE and CS legacy is performed into the VANC entity and is completely transparent to the user and the network.

The downside of VoLGA is that it has not been adopted by 3GPP and still requires all visited networks to support VoLGA in order to provide the roaming capability.

2.3 Third Party Software For VoLTE

Another solution that permits voice over LTE is enabled by using a third party software such as Viber or Skype, for providing calls over the LTE as presented in the figure3. No changes are required in the network [6], which makes the solution the least expensive among others VoLTE solutions. The downside of this solution is that there is no quality of service guaranteed and this solution doesn't support the CS calls.

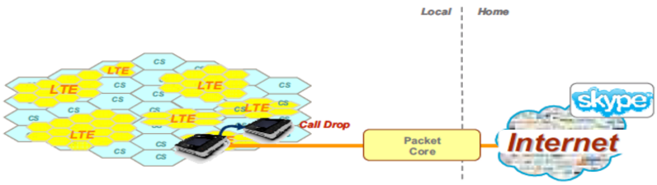

Fig. 3. Voice Over LTE via Skype

3 SINGLE RADIO VOICE CALL CONTINUITY

The Single Radio Voice Call Continuity (SRVCC) is the 3GPP target solution, based on IMS, for ensuring the voice continuity between LTE and 2/3G networks. The SRVCC requires that the UE initiates the call using IMS and the Application Server that needs to be implemented in the IMS [4].

The SRVCC uses a combination of the IMS session continuity procedure with a handover procedure. In detail, when a UE reaches the border of the LTE coverage, the E-UTRAN will send a SRVCC handover request to the MME, the latest will trigger the SRVCC procedure with the enhanced MSC (eMSC). The eMSC will perform the session transfer to the IMS and coordinates it with the CS handover to the target cell. Finally the eMSC will send the Forward Relocation Response to MME, which includes the handover command. The figure 4 and 5 presents the architecture and call flow of the SRVCC [4], [5].

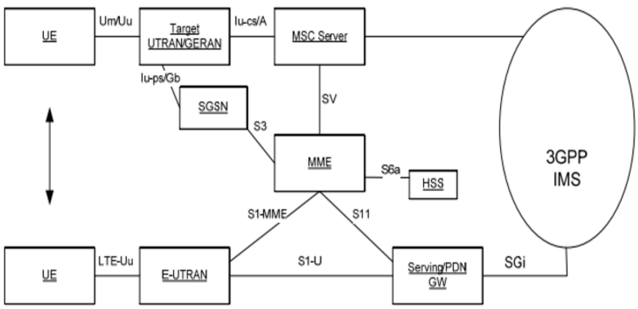

Fig. 4. SRVCC Architecture for E-UTRAN to UTRAN/GERAN

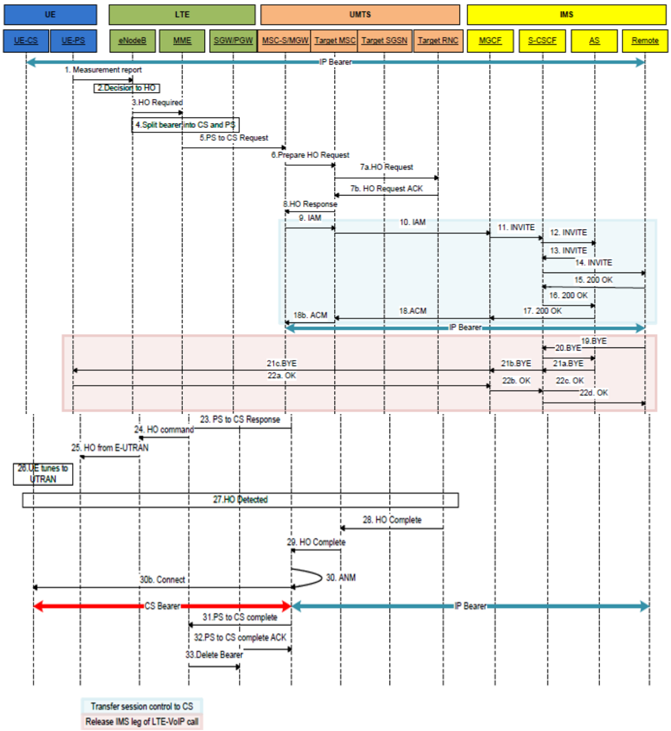

Fig. 5. SRVCC call flow from E-UTRAN to UTRAN/GERAN

4 Analytical Model for SRVCC

Many papers have studied the SRVCC handover and tried to come out with a standard model to simulate the performance of this long term solution for voice over LTE.

One of the mathematical models that have been developed is the model presented in [8], in which an analytical expression of the overall delay experienced by the call executing the handover is been presented. This delay was based on the sum of the delay caused by the radio link (RLC) and the delay caused by the remote and internet queuing which was assumed to be following the First In First Out queuing algorithm.

The results of [8] was more related to the radio link delay and has shown that the SRVCC handover interruption time will be reduce under a given Block Error Rate (BLER) with an increased data rate as presented in the figure6.

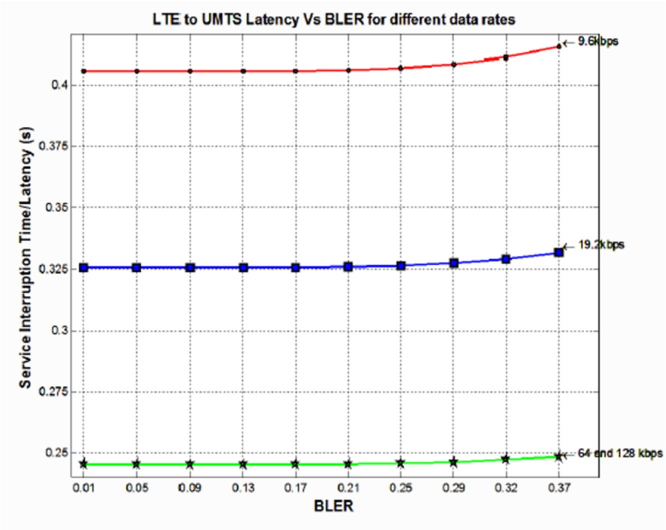

Fig. 6. LTE to UMTS SRVCC service interruption time Vs BLER

In this paper, we will be reevaluating the model while assuming that the Quality of Service is applied in the whole network since it is crucial for LTE to implement the QoS in order to serve fairly the different traffics that flow through the network.

We estimate that applying a more advanced queuing algorithm will minimize the delay of the SRVCC packets in the network nodes and there for reduce considerably the overall delay experienced by SRVCC handover which will enhance the performance of the handover.

In order to evaluate the performance of the SRVCC under priority queuing, it is necessary to analyze the interruption time experienced by the ongoing call. This delay, and based on the [9], can be split into three parts: Radio link delay, the network queuing delay and the remote network queuing delay. These delays can be represented by be mathematical models since they are following a unique mathematical behavior.

1. The radio link delay:

Based on several papers, the radio delay can be measured by analyzing the Radio Link Control (RLC). Assuming the same RLC model in LTE as UMTS, we can represent the delay in the radio part by the following equation1 [8]:

$$T_{RLC} = T_{Iub} + (k-1)TTI + \frac{k(P_s - (1-p))}{P_s^2} * \left\{ \sum_{j}^{n} \sum_{i}^{j} \left[P(C_{ij}) \left(2jT_{Iub} + \left(\frac{j(j+1)}{2} + i \right) * TTI \right) \right] \right\}$$

Equation1: Radio Delay

Where:

K : number of frames to be transmitted
N: number of RLC retransmissions
Ps: probability of receiving RLC frame successfully after n transmission
P: probability of RLC frame received erroneously
TIub : latency of the Iub interface
TTI: transmission time interval at eNodeB
P(Cij): the first correctly received frame at destination

2. The network node and remote queuing delay:

The delay caused by the nodes queues relies heavily on the algorithm applied in the system. Many algorithms has been developed and globally used such as FIFO, Priority Queuing (PQ) and Weight Round Robin PQ.

In this research and since LTE is an ALL-IP network that must apply queuing algorithm that guarantee the Quality of Service (QoS) in the network in delivering the traffic, we, first, assume that PQ algorithm has been applied in order to priorize packet coming from the handover process and therefore reducing their treatment delay in the queue.

In order to estimate the average waiting time (delay) in the queues for both the network queuing delay and for the remote queuing delay, we assume that the packets priority is p and arriving with a Poisson distribution with parameter λ_p and their length has an exponential distribution with an average service time $\overline{x_p}$.

We can then calculate the average waiting time W_p of priority p packets in the following way [10]:

$$W_p = \frac{W_0}{(1-\sigma_p)(1-\sigma_{p+1})}$$

Equation2: Average waiting time for PQ

Where W_0 is the average delay caused by a packet, which is already serviced and calculated as in:

$$W_0 = \sum_{i=1}^{P} \frac{\lambda_i \overline{x^2}}{2}$$

Where $\overline{x^2}$ is the second moment of service time.

And

$$\sigma_p = \sum_{i=p}^{P} \rho_i = \sum_{i=p}^{P} \lambda_i \overline{x_i}$$

After identifying the delay in each part of the involved part of the network, we will deduct the expression of the whole interruption delay experienced by a user that executes the SRVCC handover.

Based on the signal flow in the SRVCC in the figure5, we can notice that the service interruption start when the call is released in order to establish a new connection with the 3G network. The figure7 [8] represent a zoomed part of the signal flow during the interruption of the service.

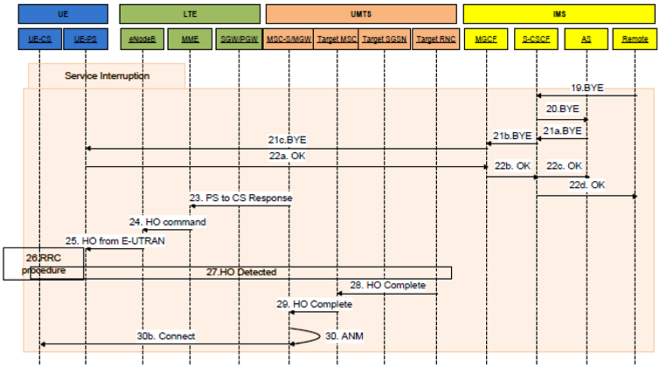

Fig. 7. Service Interruption

Based on the figure and message flow in each node and radio link, we can write the equation that represent the overall interruption delay as follow:

$$Dealy_{SRVCC} = Delay_{Remote} + Delay_{Internet} + 2 * Delay_{AS} + 3Delay_{CSCF} + 2Delay_{MGCF} + Delay_{UE} + 3Delay_{MSC} + Delay_{MME} + Delay_{eNodeB} + Delay_{RRC} + Dleay_{RNC}$$

Equation3: Overall SRVCC interruption time

Each value of the equation3 will be replaced by its expression that has already been presented in the section1 and 2. The equation3 will be used to estimate the delay of the interruption of service of the ongoing call until the SRVCC handover has reestablish the circuit with the new cell. The delay will be evaluated under different parameters values in order to study the behavior of the SRVCC performance.

The simulation results and performances evaluation for different priority queuing will be presented in separate paper in the near future.

5 Conclusion

Providing voice calls in LTE has been a challenge and a necessity for the operators since LTE is designed for only PS and the significant incomes generated by the voice calls. Many temporary solutions has been proposed and used, however the operators still seeking for implementing the long term solution SRVCC that will enable the transmission of voice over LTE and supporting the CS over LTE. This solution has not been studied in depth in order to evaluate it performance. This paper has presented the different temporarily solution that are up to date and has focused on the long term solution while proposing a new analytical model that could be used to evaluate the performance of SRVCC. This model will be simulated via Matlab and that for different priority queuing in order to evaluate the performance of SRVCC under these conditions.

References

[1] J. E. Vargas Bautista, S. Sawhney, M. Shukair, I. Singh, V. K. Govindaraju, and S. Sarkar, "Performance of Fallback from LTE to UMTS" Qualcomm Corporate Engineering. IEEE Communications Magazine, Vol. 51, Issue: 9, pp. 136-143, 2013.

[2] CS fallback in Evolved Packet System; Technical Specification 3GPP TS 23.272; 2010.

[3] Voice over LTE with Generic Access, Technical Specification VoLGA Stage 2; 2010.

[4] Single Radio Voice Call Continuity; Technical Specification 3GPP TS 23.216; 2008.

[5] Single Radio Voice Call Continuity; Technical Specification 3GPP TR 23.856; 2010.

[6] S.Gavrilovic, "Standard Based Solutions for Voice SMS Services over LTE"; IEEE/MIPRO; pp. 334-339, 2010.

[7] Cristopher Cox, "Introduction to LTE", Wiley, 2012.

[8] Namakoye, J.; Van Olst, R "A Performance evaluation of a voice call handover scheme between LTE and UMTS" IEEE Conference Publications, 2011

[9] Kim, Kyungmin. "A seamless voice call handover scheme for next generation cellular network". Shanghai : IEEE, 2009. IEEE 15th Asia-Pacific Conference on Communications, 2009.

[10] Tomáš Balogh, Martin Medvecký, "Comparison of Priority Queuing Based Scheduling Algorithms" ElektroRevue, 2010

[11] Enderle N., Lagrange X. "Radio link control-acknowledged mode protocol performance modeling in UMTS". IEEE, 4th International Workshop on Mobile and Wireless Communications Network, 2002