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Performance Analysis and Deployment of VoLTE Mechanisms over 3GPP LTE-based Networks

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***Abstract*— Long Term Evolution based networks lack native support for Circuit Switched (CS) services. The Evolved Packet System (EPS) which includes the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC) is a purely all-IP packet system. This introduces the problem of how to provide voice call support when a user is within an LTE network and how to ensure voice service continuity when the user moves out of LTE coverage area. Different technologies have been proposed for the purpose of providing a voice to LTE users and to ensure the service continues outside LTE networks. The aim of this paper is to analyze and evaluate the overall performance of these technologies along with Single Radio Voice Call Continuity (SRVCC) Inter-RAT handover to Universal Terrestrial Radio Access Networks/ GSM-EDGE radio access Networks (UTRAN/GERAN). The possible solutions for providing voice call and service continuity over LTE-based networks are Circuit Switched Fall Back (CSFB), Voice over LTE via Generic Access (VoLGA), Voice over LTE (VoLTE) based on IMS/MMTel with SRVCC and Over The Top (OTT) services like Skype. This paper focuses mainly on the 3GPP standard solutions to implement voice over LTE. The paper compares various aspects of these solutions and suggests a possible roadmap that mobile operators can adopt to provide seamless voice over LTE.**

***Index Terms*— VoLTE, E-UTRAN, SRVCC Inter-RAT Handover, CSFB and VoLGA**

I. INTRODUCTION

THE Third Generation Partnership Project (3GPP) has developed a new technology called 3GPP Long Term Evolution (LTE) in Release 8 (R8) technical specification [1]. 3GPP LTE aims to improve the third generation (3G) Universal Mobile Telecommunication System (UMTS) technology to meet the International Mobile Telecommunications-Advanced (IMT-A) requirements determined by ITU [2]. Some of the agreed requirements of LTE are a significant increase in data rates to 100 Mbps (downlink) and 50 Mbps (uplink); a scalable bandwidth and a reduced latency [3]. Moreover, a flat all-IP network architecture has been adopted. However, the price of this is high with the Evolved Packet Core network (EPC) only support Packet Switched (PS) services [1]. The EPC lacks native CS services support, including voice which is considered as the main revenue for mobile operators. This is different from most of UTRAN/GERAN wireless networks such as GSM/GPRS and WCDMA, which support both CS and PS services [2]. A user always expects voice as a basic service provided by the network operator so this raises the question of how to provide voice calls to LTE users and how to ensure service continuity during movement from one wireless network to another. This paper discusses mainly two technologies standardized by 3GPP to provide voice service, Circuit Switched Fall Back (CSFB) and Voice over LTE (VoLTE) based on IP Multimedia Subsystem/ Multi Media Telephony (IMS/MMTel) with the vertical Inter- Radio Access Technology (Inter-RAT) handover namely Single Radio Voice Call Continuity (SRVCC) for service continuity.

Non-3GPP solutions such as VoLGA and OTT/UMA are investigated in this paper briefly. The contribution of this paper is in the analysis and comparison between all these mechanisms based on different aspects such as QoS, cost of deployment is introduced clearly. Moreover, suggests a possible roadmap that mobile operators can adopt to provide seamless voice over LTE.It is important to make sure that these proposed solutions are efficient enough to provide a voice to the end user and does SRVCC provide a seamless handover between UTRAN/GERAN and Evolved UMTS Terrestrial Radio Access Network (E-UTRAN)? These questions bring us to the main research question of this paper: which technology shall be used to provide voice and service continuity over LTE. In order to answer these questions, a technical and performance analysis based on operator's technical implications and on previous aspects are carried out and recommendations are made.

The rest of this paper is organized as follows. Section II gives a description of the VoLTE technology. Section III explains the Inter-RAT SRVCC handover from LTE to UTRAN/GERAN. Section IV describes the CSFB. Section V explains and analyses briefly non 3GPP mechanisms. Section VI gives a detail analysis and performance analysis for the 3GPP and non 3GPP mechanisms and section VII concludes the work.

II. VOLTE BASED ON IMS/MMTEL

Providing voice services are considered fundamental to the wireless mobile operators. IP Multimedia Subsystem IMS [4] with MMTel are the key to make this possible and provide a required telephony system to LTE [5]. In VoLTE technology, a software upgrade is required to the LTE network and its PS core network (EPC). These voice services use the same Mobile Subscriber Integrated Services Digital Network Number (MSISDN) to provide High Definition (HD) voice calls and other Circuit Switched (CS) services. The first VoLTE service was launched commercially in Korea and US using Ericsson products and services in August 2012 [6]. VoLTE uses a Quality of Service Class Indicator value equal to one (QCI=1) in Guaranteed Bit Rate (GBR) resource type and the conversational QoS class for either originating or terminating a voice call. This guarantees the required QoS for VoLTE service. The procedure for the UE to originate a voice call in a roaming scenario (Fig. 1) Is started when the UE sends the SIP INVITE request [7], containing an initial SDP to the Proxy-Call Session Control Function (P-CSCF) determined via the CSCF discovery mechanism. From the IMS registration procedure, P-CSCF remembers the next hop CSCF for this UE. In this case it forwards the INVITE to the S-CSCF in the home network. Serving- CSCF (S-CSCF) validates the service profile, if a Globally Routable UA URI (GRUU) is received as the contact. S-CSCF forwards the request, as specified by the Specification series procedures. The media stream capabilities of the destination are returned along the signalling path, per the S-S procedures. S-CSCF forwards the Offer Response message to P-CSCF. P-CSCF authorizes the resources necessary for this session. P-CSCF forwards the Offer Response message to the originating endpoint. UE decide the offered set of media streams and sends the Response Confirmation to P-CSCF. The Response Confirmation may also contain SDP. The UE initiates resource reservation procedures for the offered media. Otherwise, the IP Connectivity Access Network (IP-CAN) initiates the reservation of required resources.

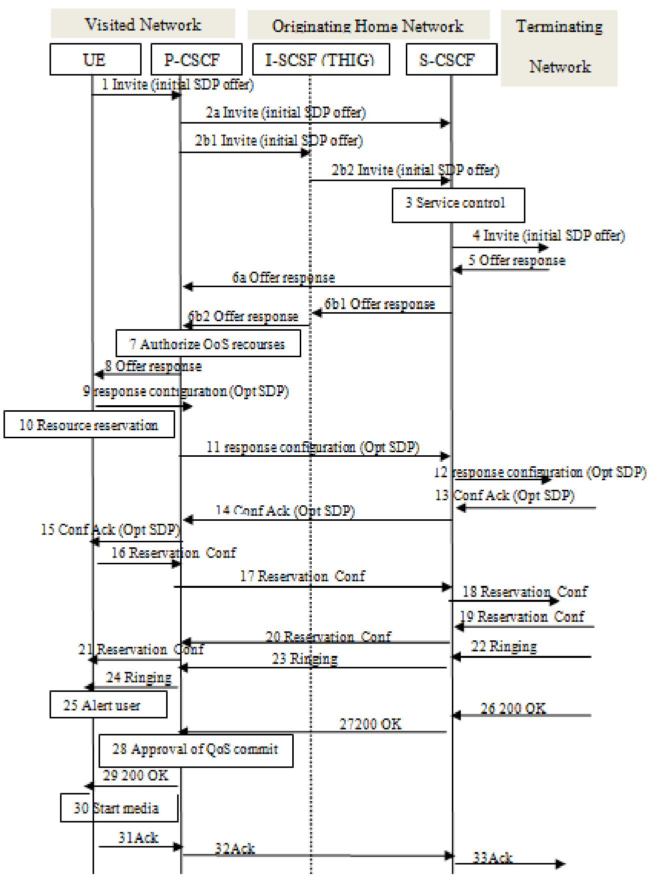

Figure 1: Mobile Origination Procedure/roaming- source 3GPP

P-CSCF forwards this message to S-CSCF. S-CSCF forwards this message to the terminating endpoint, as per the S-S procedure. The terminating end point responds to the originating end with an acknowledgement. When the resource reservation is completed, UE sends the successful Resource Reservation message to the terminating endpoint, via the signalling path established by the INVITE message. The message is sent first to P-CSCF. The terminating end point responds to the originating end when a successful resource reservation has occurred. If the SDP has changed, the P-CSCF again authorizes which resources are allowed to be used. The destination UE may optionally perform alerting. If so, it signals this to the originating party by a provisional response indicating Ringing. UE indicates to the originating user that the destination is ringing.

When the destination party answers, the terminating endpoint sends a SIP 200-OK final response along the signalling path to the originating end. P-CSCF passes the 200-OK response back to UE . The UE starts the media flow(s) for this session. The UE responds to the 200 OK with an ACK message which is sent to P-CSCF and passed along the signalling path to the terminating end. Voice over IMS uses the Adaptive Multi Rate (AMR) speech codec with all eight modes, with a baseline profile supporting AMR narrow band. Use of AMR wide band is recommended according to the IMS profile in the GSMA document in [8]. According to 3GPP technical specification in [5], IMS is an access independent based on the Session Initiation Protocol (SIP) defined by the Internet Engineering Task Force (IETF) to support voice and other multimedia services in LTE networks [7]. The reference architecture of IMS is illustrated in Fig. 2. IMS provides a complete solution to handling voice over all-IP wireless networks. VoLTE is one of the main and important roles of IMS. This is the reason why the GSM Association (GSMA) announced that it will consider IMS as a major solution in the one voice profile recommendations in 2010 [8].

The first step of UE registration to start a voice call is an IMS registration (Fig. 3). Then the UE obtains a required bearer to complete the call followed by IP address allocation to be known by other users. In order to secure the connection during the session between UE and the P-CSCF based on SIP protocol, IPSec is used.

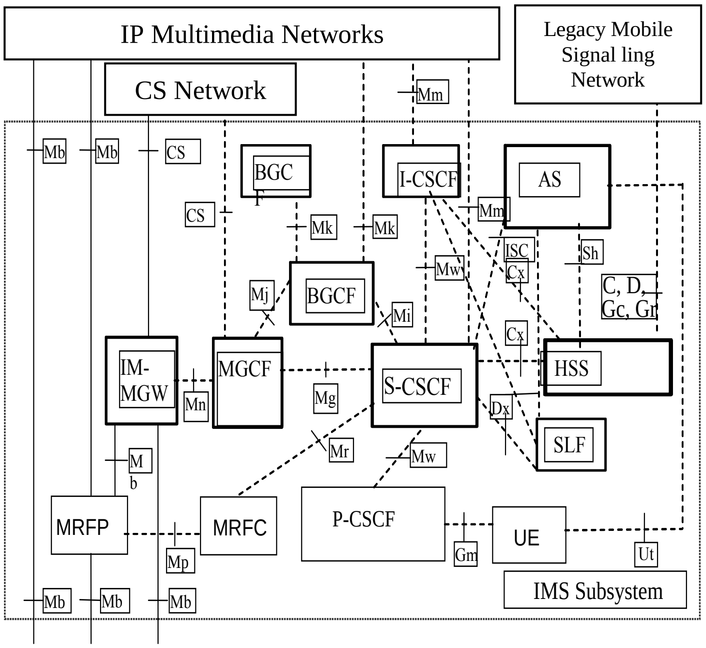

Figure 2: The IMS Reference Architecture- source 3GPP

MMTel is a service set in the IMS standard architecture that defines both Network to Network Interface (NNI) and User to Network Interface (UNI) [9]. It offers real time multimedia services based on IMS and allows users to use voice and other services in communications. One of the major roles of MMTel is to provide a minimum performance voice and video which support the 3GPP codecs. MMTel originated in 3GPP Release 7 with many of the enhancements in the subsequent releases.

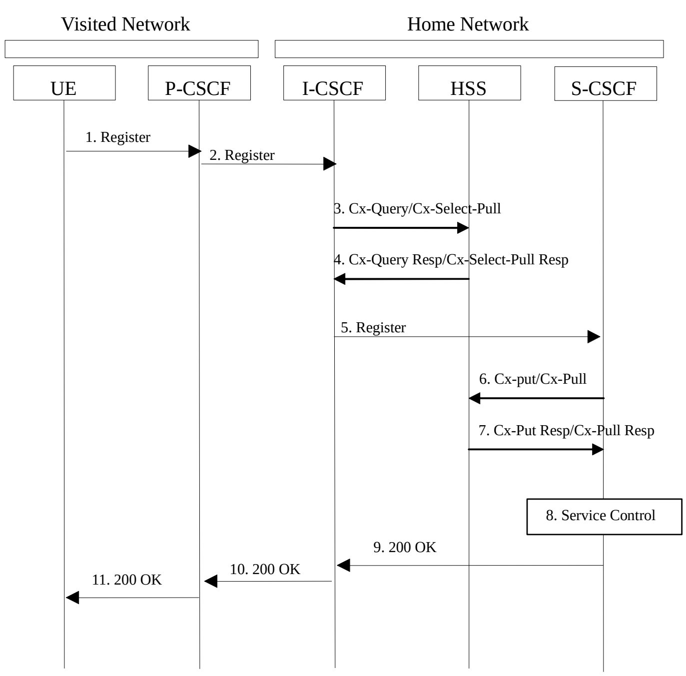

Figure 3: Initial IMS Registration- source 3GPP

III. SINGLE RADIO VOICE CALL CONTINUITY

VoLTE is a solution to provide voice calls based on IMS when the UE is in an LTE network area. However, when the UE moves out of the LTE coverage area, what will happen? Theoretically the call should be dropped directly, which will impact on the user experience. However, suppose there is radio coverage for another Radio Access Technology (RAT) such as UMTS available at that time, which is the first motivation for developing SRVCC technology. The other motivation relates to the spotty LTE network coverage which is unlikely to be available nationwide or in rural areas during the initial deployment phase. Therefore, the continuity of the voice service is a high priority for mobile operators when they are deploying their LTE networks with IMS [10]. SRVCC is an efficient Inter-RAT hard handover technology to provide guaranteed voice call continuity to the subscribers moving from an LTE PS network to a legacy CS wireless network such as UTRAN/GERAN (see Fig. 4). SRVCC supports service continuity to different kinds of legacy networks so the procedure of this handover will vary depending on the target wireless network.

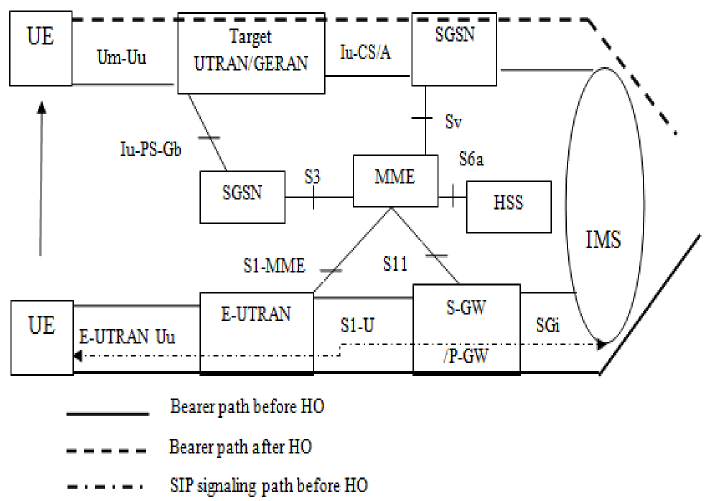

Figure 4: SRVCC for E-UTRAN to 3GPP GERAN/UTRAN- source 3GPP

SRVCC was standardized in 3GPP Release 8 [10] with many enhancements added later such as supporting emergency calls continuity in [11], supporting mid-call feature and alerting phase in [12]. Furthermore, supporting video call continuity with the voice call handover ability from UTRAN/GERAN to E-UTRAN was also introduced in [13]. The prerequisite for SRVCC is that the User Equipment (UE) should have initiated a voice call using IMS with an Application Server (AS) for session transfer in the LTE coverage area and then moved to the new RAT coverage area. SRVCC support UE and IMS service continuity capability with only a single radio access by the UE at a given time. There is no need for multi RAT capability for UE in SRVCC. In case the target legacy network is UTRAN or GERAN (Fig. 5) Then the MSC server reserves the necessary resources in the CS side to prepare the handover procedure [10]. In parallel, the Mobility Management Entity (MME) triggers the session transfer procedure at the Services Centralization and Continuity Application Server (SCC AS). The MME connects to the MSC server via Sv interface; the MME uses this interface to start relocation and session transfer. SCC AS needs to enable IMS Centralized Services (ICS) which are used to set up and control IMS sessions using CS barriers that are established between the UE and the SCC AS.

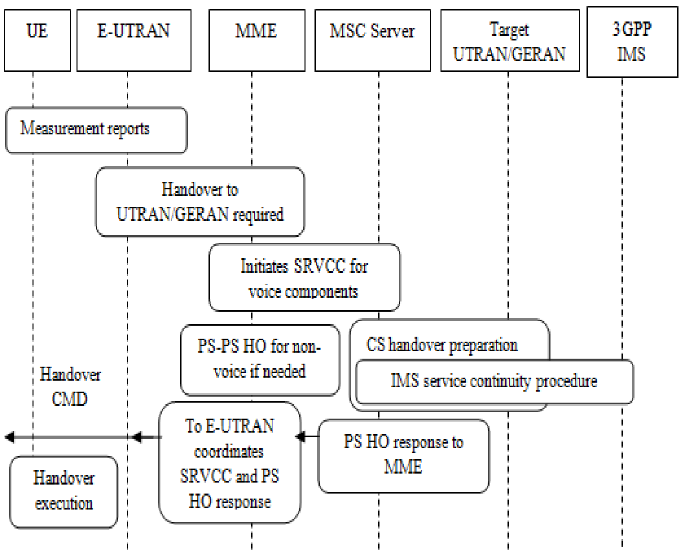

Figure 5: Network Level Procedure for SRVCC from E-UTRAN to GERAN/UTRAN- source 3GPP

Fig. 6 illustrates the SRVCC CS handover procedure from E-UTRAN to GERAN [10]. The handover starts when the UE sends measurement reports to E-UTRAN. Then E-UTRAN (based on measurement reports) decides to initiate the SRVCC handover to GERAN and sends the handover required message (Target ID, generic source to target transparent container, SRVCC HO Indication) to the MME. The MME triggers the SRVCC procedure for the voice bearer towards the MSC server. The MSC server initiates the session transfer procedure to IMS and coordinates it with the CS handover procedure to the GERAN. Then a standard IMS service continuity procedure is applied for the execution of the session transfer. The MSC server sends a PS-CS handover response with the STN-SR to MME including CS handover command information for the UE to access GERAN. The MME sends a handover command message to E-UTRAN, which includes the information about the voice bearer. This message is encapsulated within the mobility information from the E - UTRAN command and sent to UE. Finally, UE switches to GERAN and resumes its voice call.

SRVCC also has the ability to hand over non-voice sessions. In this case a PS bearer splitting function in MME is responsible for splitting voice and non-voice bearer [13]. If the target is UTRAN or GERAN with Dual Transfer Mode (DTM) capability, E-UTRAN sends the required handover message. The message includes in this case the Target ID, generic source to Target Transparent Container (TTC), additional source to TTC and SRVCC HO indication to MME. From this message and the SRVCC hand over identification, the MME identifies that this handover is for both PS and CS.

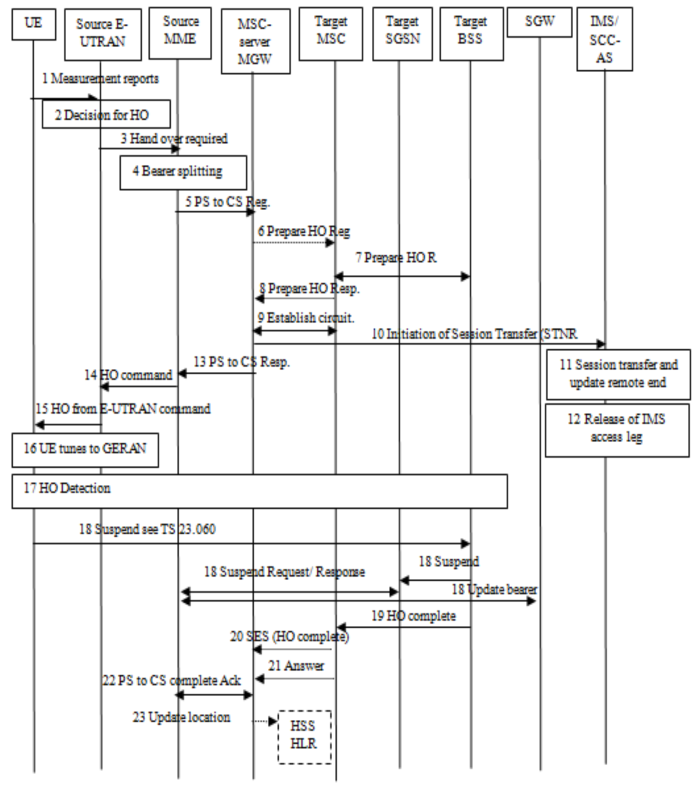

Figure 6: SRVCC CS Handover Procedure from E-UTRAN to GERAN without DTM- source 3GPP

IV. CIRCUIT SWITCHED FALLBACK (CSFB)

CSFB [14] is a bridging technology between the LTE PS and legacy CS wireless networks to obtain CS services (Fig. 7). The Next Generation Mobile Network's alliance (NGMN) has recommended CSFB to enable non IMS roaming subscribers to use both PS and CS voice services in legacy CS networks. The precondition in CSFB is the LTE coverage must overlap with either UTRAN or GERAN. CSFB was specified in 3GPP technical specification in [14], with further enhancements in release 9 and beyond. A number of different CSFB mechanisms are available depending on the target radio the UE falls back to, such as UTRAN/GERAN and non 3GPP networks. CSFB cannot support UTRAN/GERAN and non 3GPP networks simultaneously in the same Public Land Mobile Network (PLMN) even if the UE would support them [15]. The UE uses the same Mobile Subscriber Integrated Services Digital Network (MSISDN) number in LTE and CS networks. An additional functionality is added to the S3 reference point between the MME and the SGSN. The important interface in the CSFB mechanism is SGs between MSC server and MME (Fig. 8). This interface is based on Gs interface between MSC and SGSN and it provide almost all the functions provided by Gs. The main procedures provided by SGs are Mobility Management (MM) and paging between E-UTRAN PS domain and CS domain. In CSFB, the UE handles originating and terminating calls to CS networks according to the following procedures [14]:

A. Originating a CSFB voice call

1. The UE sends CSFB request message (extended service request) to the MME.
2. The MME responds to the UE by sending a handover command to handle required bearers towards UTRAN/GERAN and start a handover procedure.
3. This MME response indicates to eNodeB that the UE should change its radio and move to UTRAN/GERAN.
4. The eNodeB triggers Inter-RAN handover to a UTRAN/GERAN neighbor cell by sending a handover required message to the MME.
5. After the successful handover from the PS domain to the CS domain, a CS voice call using UTRAN/GERAN has established a normal voice call procedure.
6. CSFB is complete.

Note that CSFB does not cause interruption in active data sessions when the UE start a voice call. There are three different scenarios to handle these data sessions:

- These data sessions may hand over to UTRAN/GERAN and proceed depending on the characteristics of the target network. If the target network is UTRAN such as UMTS then Packet Switch handover (PSHO) is required from E-UTRAN to UMTS. However, if the target is GERAN such as GSM, there are two more possibilities. CSFB could provide data service continuity if the GSM and the UE support Dual Transfer Mode (DTM) to enable data and voice to be handled together at the same time. Otherwise, the data sessions will drop. Note that even if the target network provide data service continuity to active data sessions but the voice call finished before finishing the data sessions then it might either hand over back and continue on the LTE network or simply drop. DTM is not mandatory for CSFB to work.
- These data sessions may suspend during the voice call and start again in the LTE network.
- These data sessions may simply drop.

In all events, the user will experience clear degradation in QoS.

B. Terminating a CSFB voice call

1. The MSC/VLR in UTRAN/GERAN receives a message for a mobile terminating call.
2. From the call information the MSC/VLR identifies the corresponding MME and then it sends a paging request to that MME.
3. The MME in turn sends the paging request to the UE.
4. The UE sends a CSFB message to the MME after knowing that this call is in the CS domain.
5. eNodeB starts the handover procedure for the UE to the CS domain.
6. The UE receives the voice call after moving to UTRAN/GERAN target side.

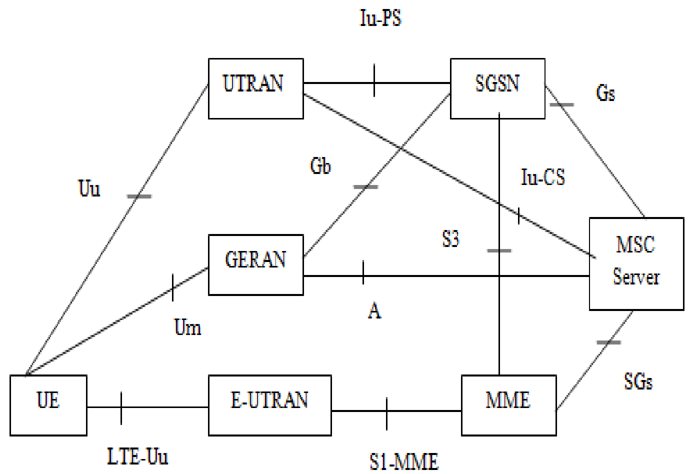

Figure 7: EPS Architecture for Circuit Switched Fallback over SGs - source 3GPP

CSFB provides CS voice calls, emergency calls and data service continuity in addition to SMS [15]. CSFB extends the life of the UTRAN/GERAN networks and their equipment by using them again to provide CS services to LTE subscribers. No network modification is required except an upgrade to the current MSC server and IMS. CSFB is convenient to use during the new LTE network deployment. Moreover, CSFB is suitable to use in the LTE roaming scenario when the visiting LTE networks do not have IMS or IMS still not fully deployed. Conversely, the MSC upgrade must be applied to all the MSCs in the network which is very costly from the operator's point of view. CSFB is signaling intensive due to the fact that the UE fallbacks to a legacy network every time it wants to originate or terminate a call. This fallback includes the Location Area Update (LAU) procedure [16] which increases the delay time during originating or terminating CSFB calls.

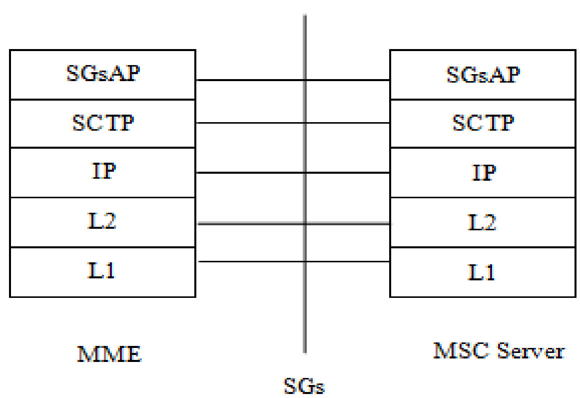

Figure 8: SGs Reference Point between MME and MSC Server- source 3GPP

V. NON 3GPP SOLUTIONS

A. OTT and UMA

1) Over The Top (OTT)

Over The Top (OTT) means to provide voice service through third party providers such as Skype or Google talk. OTT is either free of charge or very inexpensive and a simple way to provide VoIP. No changes to the LTE network or special UE capabilities are required in this option. Mobile operators might use OTT when they do not want to invest too much money on the deployment a very expensive IMS. OTT also might be used as an interim solution before deploying IMS if the operator has a plan to do that in the future. However, there are no guarantees of QoS using this solution and no service continuity when the UE moves outside the LTE coverage area. Call drop or call failure is always possible in this method [22]. It is worth mentioning that firstly, OTT is not a mobile operator solution and it is not based on cellular technology. Secondly, voice calls based on mobile networks are the main revenue for the mobile operators now and in the future. Consequently, no operator can support on this method although calls especially international might increase dramatically every day using OTT.

2) Unlicensed Mobile Access (UMA)

Similar to the previous option, there is an ongoing emergence of various radio access technologies that provide interesting technical solutions to offer VoIP through other than cellular access. They used unlicensed LAN radio access technologies such as Wi-Fi and Bluetooth to provide VoIP through what is called Unlicensed Mobile Access (UMA) [20]. There are many advantages of using UMA based calls. It minimizes the load on the cellular access networks using cost-effective technology to expand the coverage area of the cellular networks, especially indoors. Moreover, this technology, developed by the 3GPP standard body under the name GAN [18] is attractive to the wireless operators. However, the UE in this technology must have the ability to support multiple signals and must be UMA compatible because it has to switch to cellular networks when the user moves outside the WLAN and vice versa. This helps the UE to provide service continuity without interruption but it makes it very expensive and has a high battery consumption.

B. Voice Over LTE via Generic Access (VOLGA)

VoLGA is a different mechanism to provide voice and SMS over LTE networks [17]. VoLGA has defined by the VoLGA forum in 2009 based on the 3GPP Generic Access Network (GAN) specified in [18]. VoLGA connects the LTE PS network with MSC/VLR CS in UTRAN/GERAN using a gateway called VoLGA Access Network Controller (VANC). No upgrade is required to the LTE or legacy network side. The IMS is not part of the mechanism so no IMS support is required. Only a software upgrade is required to enhance the circuit to packet gateway which already exists for GAN technology. Two important interfaces are used to connect VANC with LTE and UTRAN/GERAN. Firstly, the SGi interface which is used to connect VANC with S-GW/P-GW. Secondly, the A/Iu-CS interface which is used to connect VANC with either RNC or MSC/VLR in UTRAN/GERAN wireless networks (Fig. 9). From an LTE core network point of view, VANC looks like any other IP based external node. VANC needs to contact PCRF during the call establishment via an Rx interface in order to obtain the required QoS. VoLGA provides good QoS with acceptable setup time due to the fact that no fallback is required to legacy networks. An emergency calls are supported in the last technical specification of VoLGA and further it supports SIM less emergency calling [17]. SRVCC handover could be used in VoLGA when the UE moves outside an LTE coverage area. Note that a feature called Local Breakout is used to reach VANC in the visited network [19]. However, VoLGA has not been accepted by the 3GPP standardization body yet, which is a big disadvantage of this technology. VoLGA replaces Wi-Fi and GSM/UMTS dual radio access networks in standard GAN technology with LTE and GSM/UMTS radio access networks. The procedure for a UE to originate a voice call using VoLGA is started when the UE switches ON then registers in the MME. The MME tries to retrieve subscriber data for the UE from the wireless network databases HLR/HSS through the S6a interface [19]. The UE establishes a connection to VANC, therefore, it needs a new IP and connection bearers which obtained using DHCP or it might be acquiring them from the home network. Now the UE open a secure IPSec tunnel with VANC over EPC in the LTE side using the SGi interface. VANC authenticates the UE using the authentication information retrieved from the HLR/HSS. Next, the UE registers to the MSC/VLR through VANC and an IPSec secure tunnel.

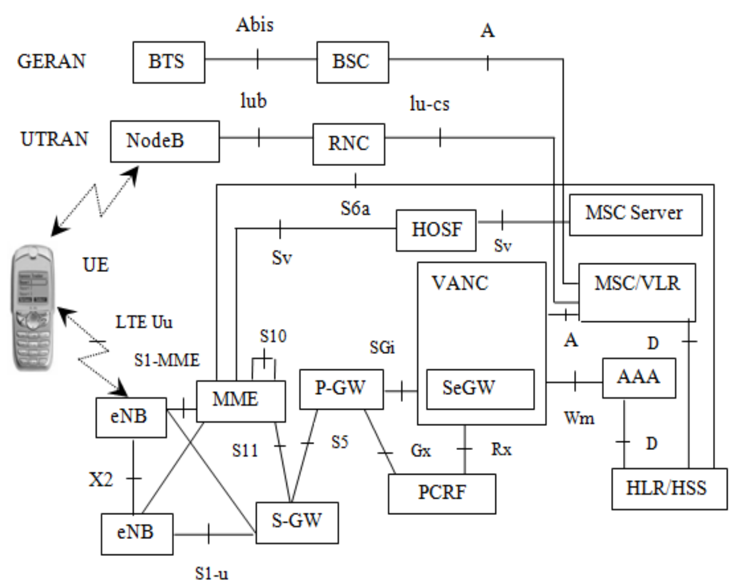

Figure 9: VoLGA Architecture to GERAN/UTRAN with connection to PCRF to ensure QoS/ none roaming- source VOLGA Forum

VI. PERFORMANCE ANALYSIS

An agreement between wireless network operators to implement a standard technology to provide voice over LTE networks is highly recommended. The first step would be an agreement on one voice profile between six operators, three mobile service providers and three handset manufacturers in 2009 [21]. The GSMA adopted this approach in 2010 and one voice profile was announced later to the public at Mobile World Congress (MWG) on the same year. This agreement needs to be extended to more operators, vendors and handset manufacturers to avoid variation between the techniques that reflect negatively on the overall performance. In this section, an analysis between VoLTE technologies based on different aspects is illustrated. Different operators have different LTE deployment plans and a choice must be made by mobile operators to support one of these mechanisms. This decision is based on their network characteristics and other issues related to deployment cost and how much the mobile operator decided to invest.

- ***Cost of Deployment***: Implementing a cost-effective solution to provide VoLTE is important from the operator's point of view. Therefore, it is recommended mobile operators have a clear deployment plan to deploy an interim or long term solution. CSFB is an expensive option, especially if it is considered as a long term option to provide VoLTE. This is because it needs an upgrade to all MSC servers which is a very costly process. VoLGA is cheaper than CSFB; this option needs GAN and VANC to deploy in order to provide voice over LTE using the legacy wireless network. As long as there is no need for any modification to the UE and for the LTE or legacy network, VoLGA would be an inexpensive option. OTT is a very inexpensive option. No investment by network operators is required; no modification in the UE or the network architecture is needed. VoLTE based on IMS/MMTel is an expensive option. Deploying IMS is very expensive and this is the reason why most of the mobile operators either use an interim option to provide VoLTE like VoLGA or they deploy IMS only in the limited areas such as urban areas.
- ***Quality of Service and User Experience***: Providing the required QoS and user experience is a key factor in the roll-out of any new technology. Different mechanisms offer different QoS, for example, in CSFB users face degradation in data and voice QoS when PS to CS handover is needed. Moreover, CSFB includes additional delay for call setup and it is very signalling intensive in addition to switching to legacy networks for each originating and terminating call. VoLGA connects to the PCRF to provide required QoS and no call setup delay while OTT does not provide any guarantees of QoS. In OTT, users may suffer call drop in the roaming scenario when they move outside an LTE coverage area. VoLTE provides carrier-grade QoS and using IMS allows mobile operators to support High Definition (HD) voice calls without extra delay.
- ***SRVCC Handover*:** Only VoLGA and VoLTE can use SRVCC Inter-RAT handover to provide service continuity to LTE users when they move outside an LTE coverage area.
- ***Support for simultaneous PS and CS*:** All of the VoLTE solutions have simultaneous support for data and voice, except for CSFB when the target is GSM without Dual Transfer Mode (DTM).
- ***Emergency Call*:** Handling emergency calls to the mobile subscribers is vital. Mobile operators have to provide this option to their subscribers after deploying one of the above options. All options except OTT provide emergency calls.
- ***3GPP Standardization*:** It is highly recommended for mobile operators to deploy an option which is already a 3GPP standard. Only VoLTE and CSFB has this 3GPP standardization. There is a lot of debate about why VoLGA does not have 3GPP standardization although it is used by many famous mobile vendors such as Huawei and Alcatel-Lucent [23].
- ***UE Capabilities*:** Mobile operators are always looking for a technology that does not require any special UE features which consequently make it expensive and affects the end user. Volt utilizing CSFB requires a UE with the dual radio capability [14] which makes the UE expensive. In OTT, there are no special requirements for the UE except having the ability to access the Internet. For VoLGA, this mechanism requires a GAN based dual mode UE which will include extra battery consumption. VoLTE based on IMS requires a UE with VoLTE and SRVCC capabilities [10].
- ***Deployment Plan*:** Different operators have different requirements and situations which affect their final decisions to deploy one of the technologies. Operators which have legacy networks and do not want to deploy IMS and do not have a plan to deploy it in the future, and do not have GAN, then CSFB might be a good choice as a long term solution. CSFB could be useful to provide voice over LTE in a roaming scenario when the visiting LTE networks do not have IMS or IMS is still not fully deployed. Similarly, operators need to maintain their legacy networks in addition to GAN and VANC when they decide to consider VoLGA, so it could be also used it as a long term solution. OTT can be used anytime, anywhere by any operator offers no guaranteed QoS, but needs no support or investment. No legacy networks are required in case of VoLTE based on IMS/MMTel. If the mobile operators have their legacy networks and have a plan to deploy IMS or do not have legacy but have a plan to deploy IMS in the future, then VoLTE would be best long term solution.

VII. CONCLUSIONS

The potential solutions for providing voice call while a user is in LTE-based networks are CSFB, VoLTE with SRVCC for service continuity, VoLGA and OTT services. For CS fallback the call is always initiated in the legacy network side hence voice call continuity does not apply. For OTT, the call cannot continue if the user moves to UTRAN/GERAN networks. The solution for IMS based voice calls and VoLGA is to use SRVCC when the user starts moving to UTRAN/GERAN or other legacy networks. Due to the fact that handover mechanisms maintain network connections over different wireless technologies and network architectures. SRVCC Inter-RAT handover technique to provide service continuity when the user roams outside LTE networks has been studied and analyzed. This paper focused mainly on the standard 3GPP solutions to provide voice over LTE-based networks. Suggestions to deploy one of these technologies based on different technical criteria were clearly introduced.

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