White Paper

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Single Radio Voice Call Continuity (SRVCC) with LTE

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Overview

Long Term Evolution (LTE) is heralded as the next big thing for mobile networks. It brings in promising technologies such as semi-persistent scheduling, transmission time interval (TTI) bundling, and high performance gains on Quality of end user Experience (QoE). In the end, the primary goal of LTE is to deliver ultra-high speed mobile broadband with peak data rates over 100 Mbps. However, in practical applications LTE is facing challenges to provide the same capabilities as a 2G/3G network during the initial stages of trial deployments and operators' metered investment in broad network build out.

One of the key issues of LTE is the delivery of voice services. Voice remains the "killer application" for operators because it still accounts for a large portion of their revenue. Voice will continue to remain the dominant must-have service in the network for years, and despite the technical challenges of providing service over an all-Internet Protocol (IP) radio access network (RAN), voice is seen as a basic service by the consumer; in short, it is expected. However, voice service continuity is not guaranteed when a Voice over IP (VoIP) subscriber roams between the LTE coverage area and other wireless networks—and it is a significant challenge to deliver voice over LTE networks.

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The industry is exploring and evaluating different possibilities to overcome the LTE voice issues. During this evaluation process two options are gaining significant momentum: Circuit Switched Fall Back (CSFB) and LTE VoIP-based Single Radio Voice Call Continuity (SRVCC). The latter is widely supported in the industry and has been recommended by the LTE OneVoice Initiative, which has the support of some of the world's largest operators and network equipment providers and has been endorsed by the GSM Association (GSMA).

This paper focuses on designing the LTE-IMS (IP Multimedia Subsystem) network framework to support SRVCC with the Circuit Switched network by realizing the 3GPP Standard 23.216 V 8.6. 2009-12: Single Radio Voice Call Continuity (SRVCC) (Release 8). This architecture has great appeal to carriers with a robust IMS Core and both fixed and wireless component assets in order to facilitate a converged VoIP solution.

Why SRVCC?

Rich multimedia services with video sharing, video on demand, video telephony, video conferencing, VoIP, Push-To-Talk, broadband access to Personal Digital Assistants (PDAs) and so on are currently offered with the existing capabilities of the Universal Mobile Telecommunications System (UMTS) using High Speed Packet Access (HSPA), Evolved HSPA (HSPA+), Code Division Multiple Access (CDMA) and IMS technologies. Increasing demand for these real-time mobile data services coupled with subscribers' expectations for always-on, high-quality services is driving the need for expanded network capacity and throughput. Enter LTE.

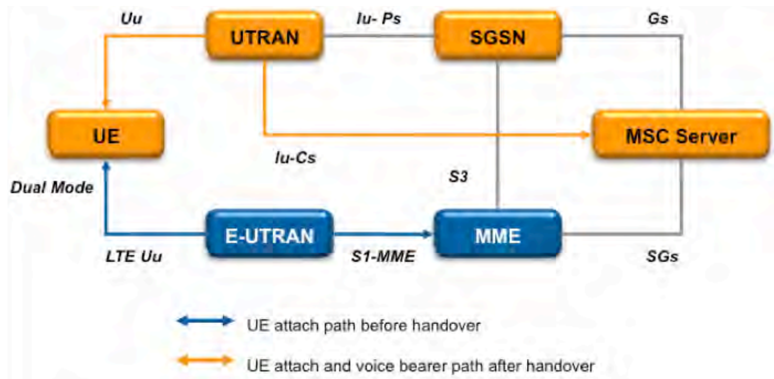

Figure 1. *EPS Reference Architecture for CSFB with UTRAN as Destination Network*

With the support of LTE's high-throughput data transmission capacity, inter-working with 3GPP and non-3GPP based networks, and all-IP core network elements, the converging services listed above can be delivered successfully. Higher bandwidth for LTE means that more resource blocks can be dispatched by the LTE system, which in turn provides higher performance gains.

Recognizing this reality, CSFB is a 3GPP-defined standard solution that requires terminals be equipped with either dual-mode/single-standby or dual-mode/dual-standby capabilities.

Figure 1 displays the reference architecture for a CSFB network using an Evolved Packet System (EPS) with the 3GPP Universal Terrestrial Radio Access Network (UTRAN).

For dual-mode/single-standby mobile phones to simultaneously use dual-network services, the Inter Working Solution (IWS) node provides on-time message access. On the other hand, dual–mode/dual-standby mobile phones require less network changes to facilitate inter-working between two networks. However, dual-mode handsets drain the battery power quickly and need complex terminal customization.

In such scenarios of converging mobile and broadband wireless access technologies, SRVCC offers LTE-IMS based voice service within the LTE coverage area, and CS-based voice service outside the LTE coverage area. Figure 2 displays the reference architecture for SRVCC using EPS to 3GPP UTRAN.

Whenever the VoIP subscriber moves out of LTE coverage, SRVCC ensures smooth handoff of voice from the LTE to the CS network, keeping upgrades of the network to a minimum. The IMS network that stores voice service link information during this time guides the target CS network to establish a link, thereby replacing the original VoIP channel.

Table 1 compares CSFB and SRVCC schemes.

The following sections describe the SRVCC from LTE to 3GPP2 1XCS and LTE to 3GPP UTRAN/GERAN CS networks.

Single Radio Voice Call Continuity From LTE

SRVCC service for LTE comes into the picture when a single radio User Equipment (UE) accessing IMS-anchored voice call services switches from the LTE network to the Circuit Switched domain—while it is able to transmit or receive on only one of these access networks at a given time. This basically removes the need for a UE to have multiple Radio Access Technology (RAT) capability.

For single-radio terminals, measurement gaps are needed to allow the UE to switch onto the CS network and complete radio measurements. Measurement gaps define the time periods when no uplink or downlink transmissions are scheduled so that the UE may perform the measurements. The Evolved NodeB (eNodeB, i.e. LTE base station) is responsible for configuring the measurement gap pattern and provides it to the UE using Radio Resource Control (RRC) dedicated signaling.

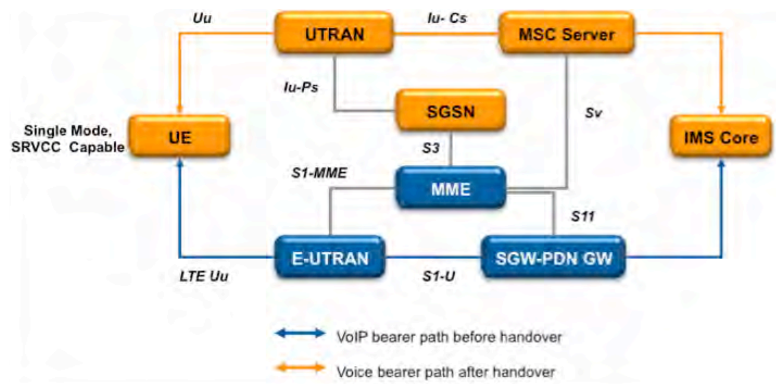

Figure 2. *EPS Reference Architecture for SRVCC to UTRAN as Destination Network*

Parameter	SRVCC	CSFB
Device/terminal capability	Single radio mode[1]	Dual-mode/single-standby or Dual-mode/dual-standby
Terminal customization	Less complex	Complex for single standby
IMS anchoring	Mandatory	Optional
Switching networks/ mobility to CS network	Only when the terminal roams out of LTE coverage area	For every mobile originating and mobile terminating voice call
Cost	Less expensive	Expensive due to increased network signaling load
Voice call setup time	Less, as time is required only when the terminal moves out of LTE coverage area	More, as the terminal needs to establish the voice call session with CS network for every access to call

Table 1.

[1] Single radio mode terminal refers to the ability of a terminal to transmit or receive on only one of the given radio access networks at a given time.

The UE assists the eNodeB by informing the network about its gap-related capabilities, at least mentioning that if it has a dual or single receiver. This capability is transferred along with the other UE capabilities. The UE accessing the SRVCC service is assumed to have IMS Service continuity capabilities with single radio access only.

SRVCC from LTE to 3GPP2 1XCS

In the case of VoIP, if subscribers geographically roam from LTE+CDMA to CDMA, voice calls must be switched from a VoIP to a CDMA 1x network using SRVCC technology. The existing inter-frequency/RAT gap pattern mechanism in E-UTRAN is therefore extended to support gap patterns suitable for 1xRTT measurements.

In this approach, the eNodeB is able to inter work with the 3GPP2 1XRTT MSC using the S1-MME interface with the Evolved Packet Core (EPC) MME. A new IWS node is required and is responsible for the exchange of 3GPP 1XCS signaling messages with the MME and for establishing a Circuit Switched session when the UE is in the process of switching over from the LTE network to the 3GPP 1XCS network. Generally, this is a case of inter system handover from the LTE perspective.

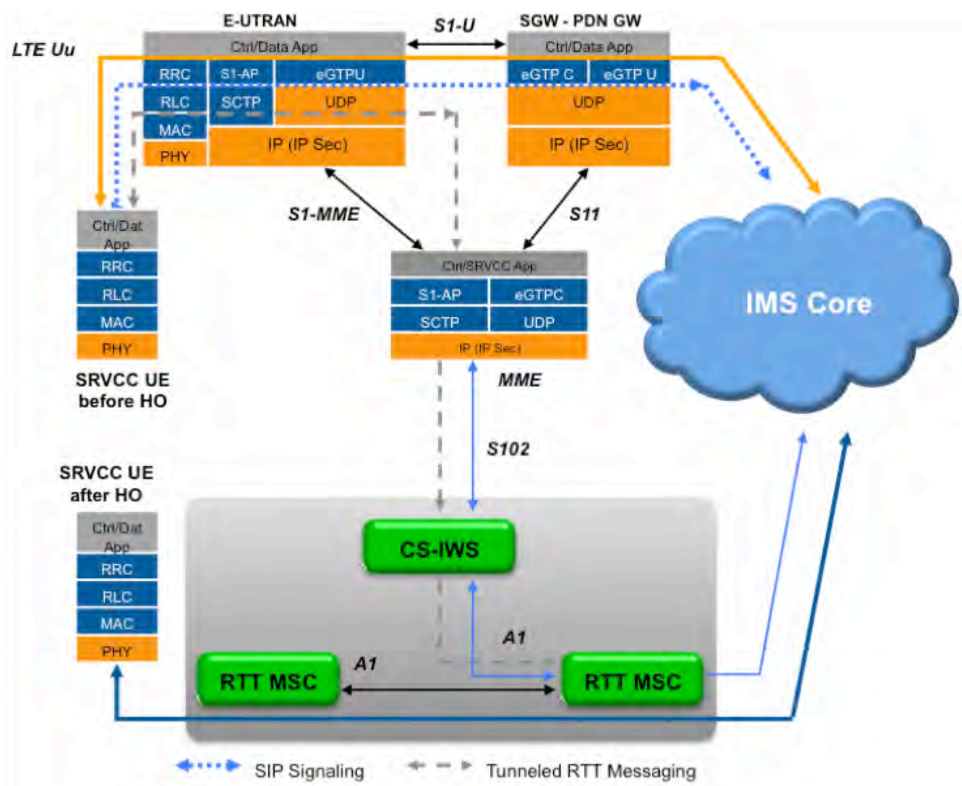

Figure 3. *SRVCC from LTE to 1x CS Network*

A new S102 reference point or interface is defined between the LTE MME node and the 3GPP 1XCS IWS node. In fact, the 3GPP 1XCS signaling messages are tunneled over this single link of S102 and thereafter tunneled through E-UTRAN/EPS tunneling messages to the UE.

Figure 3 displays the framework of SRVCC from LTE to a 3GPP2 1XCS network.

The S1-MME interface between the eNodeB and the MME ensures initiation of SRVCC service and smooth transition of these signaling messages using the S1AP signaling protocol.

Once the UE is actively attached and associated to the EPC, it interacts with the IMS core through the eNodeB, Serving Gateway (SGW) and Packet Data Network Gateway (PDN GW) by establishing a Session Initiation Protocol (SIP) signaling session. SIP therefore plays a key role along with Real time Transport Protocol (RTP), which is used for bearer plane data transfer in the SRVCC architecture.

All the SIP and RTP packets are carried through the Packet Data Control Protocol (PDCP) payload on the LTE Uu interface and are tunneled on the S1-U and the S5 interfaces using Generic Tunneling Protocol—User (GTP-U). Mapping between these packets is done at the eNodeB to interact with the Serving—PDN GW, ultimately allowing the UE access to IMS service flexibly until the handover is detected and initiated with the UE switch over.

The 3GPP2 1xCS IWS node enables a single radio UE to communicate in parallel both with the source LTE network and the target 3GPP 1XRTT system.

With this, the additional voice gap normally generated by multiple RATs being jammed into one UE is reduced by having transport of signaling establishment with target Circuit Switched access when the UE is connected to the source LTE network. However, Quality of Service (QoS) depends on the connectivity offered by the public IP network and is susceptible to IP delays and packet loss which impacts the service performance.

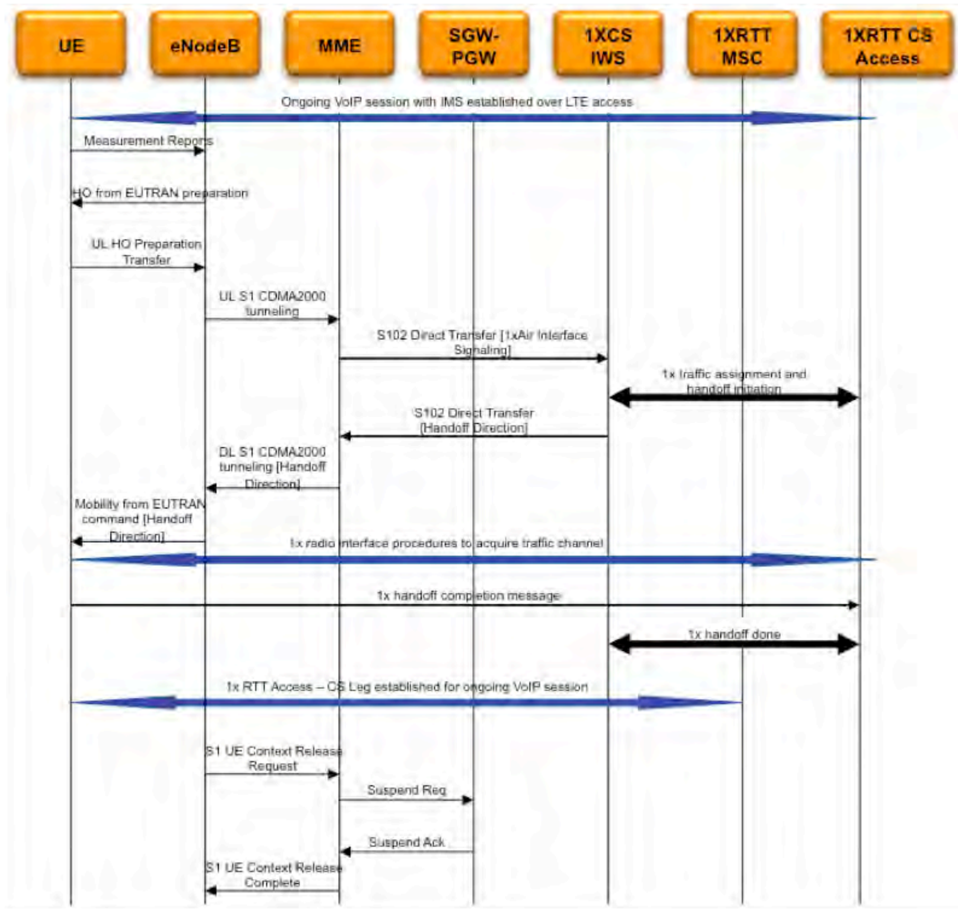

Figure 4. *SRVCC from LTE to 1x CS Voice System*

The message flow for SRVCC for a UE from LTE to a 1x CS network for VoIP IMS services is shown in Figure 4.

The entry criterion for the message flow is an ongoing VoIP session to the IMS access leg established over Evolved Packet System (EPS) access:

1. 1xCS SRVCC UE sends measurement reports to the eNodeB.
2. The E UTRAN makes a determination to initiate an inter-technology handover to cdma2000 1xRTT.
3. The E UTRAN signals the UE to perform an inter-technology handover by sending a Handover from EUTRA Preparation Request message with 3G1x Overhead Parameters.
4. The UE initiates signaling for establishment of the CS access leg by sending a UL handover preparation message containing the 1xRTT Origination message.
5. The E UTRAN sends an Uplink S1 cdma2000 Tunneling message with MEID, 1x Origination, Reference Cell ID to the MME. The eNodeB will also include CDMA2000 HO Required Indication IE to Uplink S1 CDMA2000 Tunneling message, which indicates to the MME that the handover preparation has started.
6. Upon receipt of the Uplink S1 cdma2000 Tunneling message, the MME:
   a. Separates the voice bearer from the non-voice bearers based on the QoS Class Identifier (QCI) associated with the voice bearer (QCI 1) and CDMA2000 HO Required Indication.
   b. Selects the 3GPP2 1xCS IWS based on Reference Cell ID and encapsulates the 1x Origination Message along with the MEID and RAND in a S102 Direct Transfer message (as "1x Air Interface Signaling") to the IWS, only for voice bearer.
   c. The traffic assignment is done between the IWS and RTT MSC, over the A1 interface using the signaling protocols to initiate the handoff to the 1XRTT system.

7. The traffic channel resources are established in the 1x RTT system and 3GPP2 1xCS procedures for initiation of session transfer for CS access leg are performed.
8. When the 1xRTT MSC receives a positive acknowledgment from the 1xRTT radio for traffic allocation and from the IMS for successful domain transfer, it returns an IS-41 handoff message to the IWS to send to the UE via the established signaling tunnel.
9. The 3GPP2 1xCS IWS creates a 1x message and encapsulates it in a S102 Direct Transfer message (1x, Handover indication). If the 3GPP2 access was able to allocate resources successfully, the 1x message is a 1x Handover Direction message and the handover indicator indicates successful resource allocation. Otherwise, the handover indicator indicates to the MME that handover preparation failed and the embedded 1x message indicates the failure to the UE.
10. The MME sends the 1x message and CDMA2000 HO Status IE in a Downlink S1 cdma2000 Tunneling message to the E UTRAN. The CDMA2000 HO Status IE is set according to the handover indicator received over the S102 tunnel.
11. If the CDMA2000 HO Status IE indicates successful handover preparation, the E UTRAN forwards the 1x Handoff Direction message embedded in Mobility from EUTRA Command message to the UE. This is perceived by the UE as a Handover Command message.
12. The UE now tries to acquire the traffic channel with the 1xRTT CS access as it gets aware of the traffic channel information from the cdma2000 1xRTT system.
13. The UE sends a 1xRTT handoff completion message to the 1xRTT CS access.
14. The 1xRTT CS Access sends a message to the 1xRTT MSC to indicate that the handoff is done. The traffic assignment which was done during the session/domain transfer of the CS access leg, between the 1x CS IWS and the 1xRTT MSC, is released now.
15. An ongoing voice call over the CS access leg is now established over 1xRTT access. The UE continues to transmit voice via the new access system. The voice bearer path is no longer carried by the EPC.
16. The eNodeB now initiates the release of UE context on the EPS; it sends an S1 UE Context Release Request (Cause) message to the MME. Cause indicates that S1 release procedure is due to handover from E-UTRAN to 1xRTT.
17. The MME exchanges Suspend Request and Suspend Acknowledge messages with the Serving GW. With this the S1-U bearers are released for all EPS bearers and the Guaranteed Bit Rate (GBR) bearers are deactivated by the MME. The non-GBR bearers are preserved and are marked as suspended in the S GW. Upon receipt of downlink data the S GW should not send a downlink data notification message to the MME.
18. S1 UE Context in the eNodeB and MME are now released with the normal E UTRAN/EPS procedure.

SRVCC from LTE to 3GPP UTRAN/GERAN

In this scenario, the SRVCC nodes on the target CS network and IMS network are enhanced with additional capabilities to support smooth transition of the UE from LTE to 3GPP UTRAN/GERAN based networks (e.g. 2G/3G).

1. MSC Server, is enhanced with the following features:

   a. Employed alongside the MME in LTE network through the Sv reference point.

   b. Mainly comprises the call control (CC) and mobility control parts of an MSC. The MSC Server is responsible for the control of mobile-originated and mobile-terminated CC CS Domain calls for medial channels in the CS-MGW (Media Gateway). It terminates the user-network signaling and translates it into the relevant network—network signaling, i.e., SIP Signaling in IMS and vice versa.

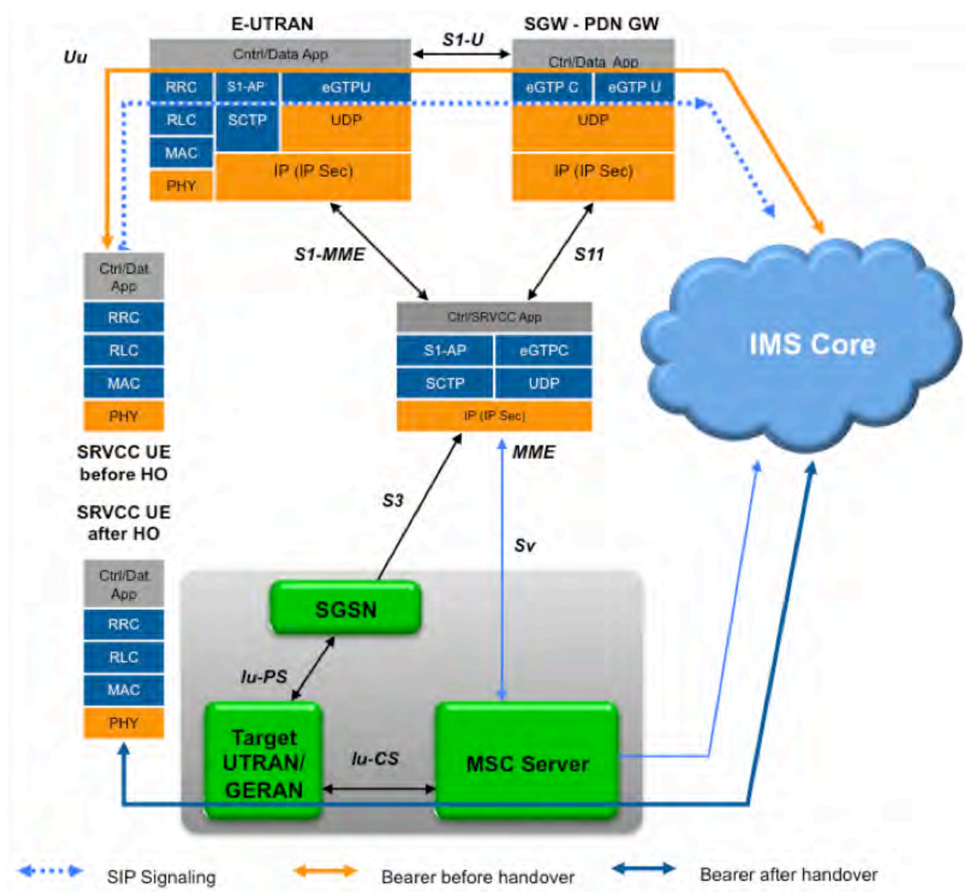

Figure 5. *SRVCC from LTE to 3GPP UTRAN/GERAN*

   c. For SRVCC, the MSC Server provides the following functions as needed:

      i. Handling the Relocation Preparation procedure requested for the voice component from the MME via the Sv interface.

      ii. Invoking the session transfer procedure from IMS to CS. This involves the access transfer at the IMS level of one or more of session signaling paths and associated media flow paths of an ongoing IMS session.

      iii. Coordinating the CS Handover and session transfer procedures.

2. Session Centralization and Continuity Application Server (SCC AS) is present in the IMS network and provides the following functionality in support of SRVCC:

   a. Required to enable IMS Centralized Services. The ICS User Agent (IUA) function furnishes SIP UA behavior on behalf of the UE for setup and control of IMS sessions using CS bearers that are established between the UE and the SCC AS.

   b. For executing and controlling the session transfers needed by the UE for its access legs anchored in IMS.

The Figure 5 displays the framework of SRVCC for a UE from LTE to a 3GPP UTRAN/GERAN CS network.

The message flow is shown in Figure 6 for SRVCC from LTE to 3GPP UTRAN/GERAN CS network and is followed by a description.

The entry criteria for the following message flow is an ongoing VoIP session over the IMS access leg established over EPS access:

1. The UE sends measurement reports to E-UTRAN.
2. Based on UE measurement reports, the source E UTRAN decides to trigger an SRVCC handover to the CS Domain.
3. Source E UTRAN sends Handover Required message having Target ID, generic Source to Target Transparent Container, SRVCC HO Indication to the source MME. The E UTRAN places the "old BSS to new BSS information IE" for the CS domain in the generic Source to Target Transparent Container. The SRVCC HO indication indicates to the MME that the target is only CS capable—hence is a SRVCC handover operation only toward the CS domain.

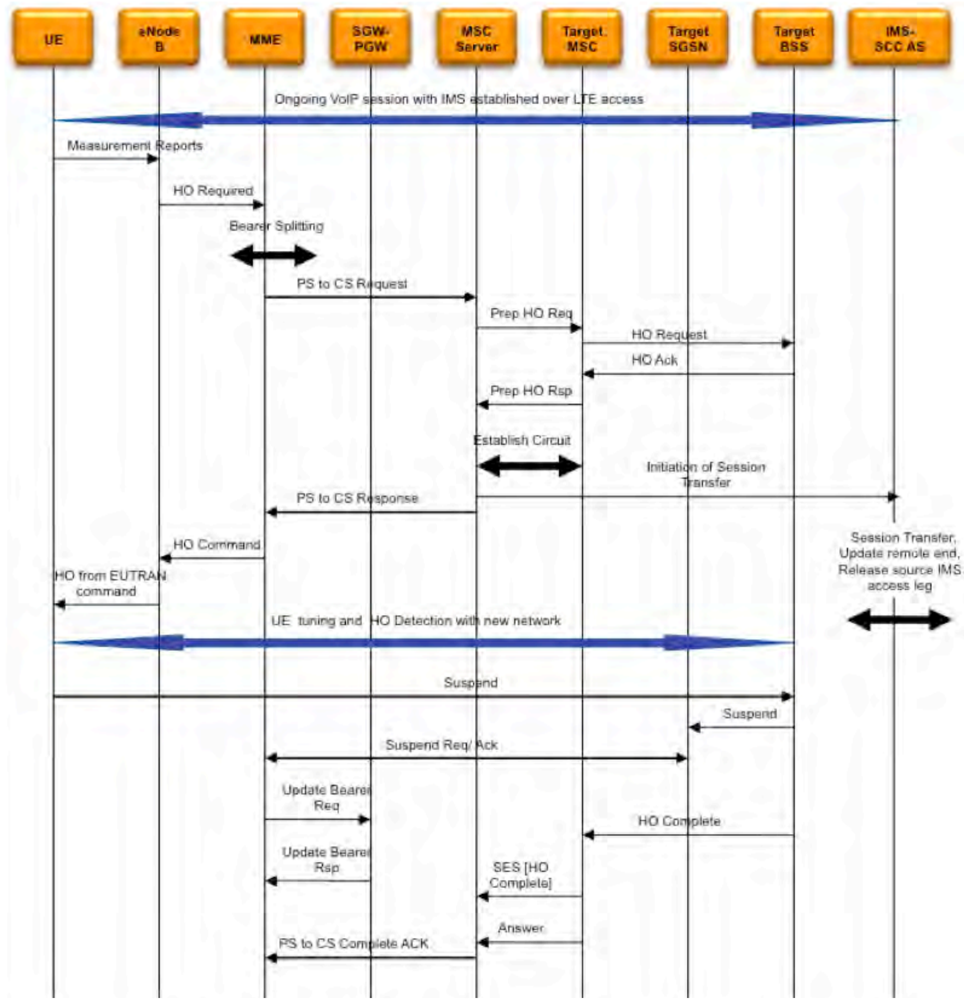

Figure 6. *Call Flow for SRVCC from E-UTRAN to GERAN without DTM Support*

4. Bearer Splitting: 3GPP has specified a QCI for VoIP services to ensure that LTE VoIP raises the quality of end user experience. Based on the QCI associated with the voice bearer (QCI 1) and the SRVCC HO indication, the source MME splits the voice bearer from the non-voice bearers and initiates the PS-CS handover procedure for the voice bearer only toward the MSC Server.
5. The MME stores the STN-SR (Session Transfer Number for SRVCC), C-MSISDN of the UE from the HSS during the UE attach procedure. The MME sends a SRVCC PS to CS Request with necessary IMSI, Target ID, STN-SR, C MSISDN and generic Source to Target Transparent Container, MM context message to the MSC Server.
6. The MSC Server interworks the PS-CS handover request with a CS inter-MSC handover request by sending a Prepare Handover Request message to the target MSC.
7. Target MSC performs resource allocation with the target BSS by exchanging Handover Request/ Acknowledge messages.
8. The target MSC sends a Prepare Handover Response message to the MSC Server.
9. Establishment of a circuit connection between the target MSC and the MGW associated with the MSC Server now occur.

10. The MSC Server initiates the Session Transfer by using the STN-SR (e.g., by sending an ISUP IAM (STN-SR) message toward the IMS). Standard IMS Service Continuity procedures are applied for execution of the Session Transfer.
11. A new access leg is established by the UE toward the SCC AS. Signaling and bearer resources are allocated in the transferring-in Access Network and the user's sessions are transferred from the transferring-out Access Network. The SCC AS executes Access Transfer procedures.
12. During the execution of the Session Transfer procedure the remote end is updated with the SDP of the CS access leg. The downlink flow of VoIP packets is switched towards the CS access leg at this point.
13. The source IMS access leg is released now.
14. The MSC Server sends a SRVCC PS to CS Response (Target to Source Transparent Container) message to the source MME.
15. The source MME sends a Handover Command (Target to Source Transparent Container) message to the source E-UTRAN. The message includes information about the voice component only.
16. The source E-UTRAN now sends a Handover from E-UTRAN Command message to the UE.
17. The UE tunes to the new CS Network; Handover Detection at the target BSS occurs. The UE sends a Handover Complete message via the target BSS to the target MSC. Since the target MSC is not the MSC Server, the Target MSC sends an SES (Handover Complete) message to the MSC Server.
18. The UE requests the target BSS to suspend the GPRS services with a Suspend message. The target BSS now asks the target SGSN to derive the old entity [which is an MME in this case] to suspend all the GPRS services by suspend message using TLLI and RAI pair which are derived from the GUTI allocated to UE. Since the SGSN [Target SGSN] that receives the Suspend message is not the one currently handling the packet data transmission, it triggers the Target SGSN to send a Suspend Request message to the MME. The MME returns a Suspend Response to the Target SGSN; the MME also starts the preservation of non-GBR bearers and the deactivation of the voice and other GBR bearers.
19. The target BSS sends a Handover Complete message to the target MSC.
20. The target MSC sends an SES (Handover Complete) message to the MSC Server. The speech circuit is through connected in the MSC Server/MGW.
21. Completion of the establishment procedure occurs with an ISUP Answer message to the MSC Server.
22. The MSC Server sends an SRVCC PS to CS Complete Notification message to the source MME, informing it that the UE has arrived on the target side. The source MME acknowledges the information by sending an SRVCC PS to CS Complete Acknowledge message to the MSC Server.

Conclusion

The popularity of VoIP applications coupled with the absolute need for operators to deliver voice over LTE is causing the SRVCC to gain momentum in the market. Despite the fact that SRVCC is apparently more complex than CSFB due to the requirement of an IMS core network (or application tier), it continues to be the choice of the LTE OneVoice initiative primarily due to the lack of need for complex and expensive dual-mode UEs. To combat the apparent cost and complexity of a full IMS network rollout, the industry is now focusing on simplifying the IMS protocols and defining a specific IMS "profile" for providing seamless LTE VoIP service.

SRVCC will play a key role in handing over the UE from LTE to other CS-based networks by utilizing high performance technology capabilities of LTE and the EPC.

Whenever there are competing solutions, different operators will likely choose to implement one or the other depending on multiple factors including existing network assets and the overall purpose for rolling out LTE. It is expected that CSFB may be used by a subset of operators as it requires less core network modification and changes can be pushed primarily onto the handset manufacturers. However, due to the efforts of the LTE OneVoice initiative and the ability to deliver SRVCC via a subset of IMS functionality, it is expected that the majority of LTE voice service will be provided via IMS-based SRVCC architecture.

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