VoLTE with SRVCC:
The second phase of voice evolution for mobile LTE devices

White Paper
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LTE Growth

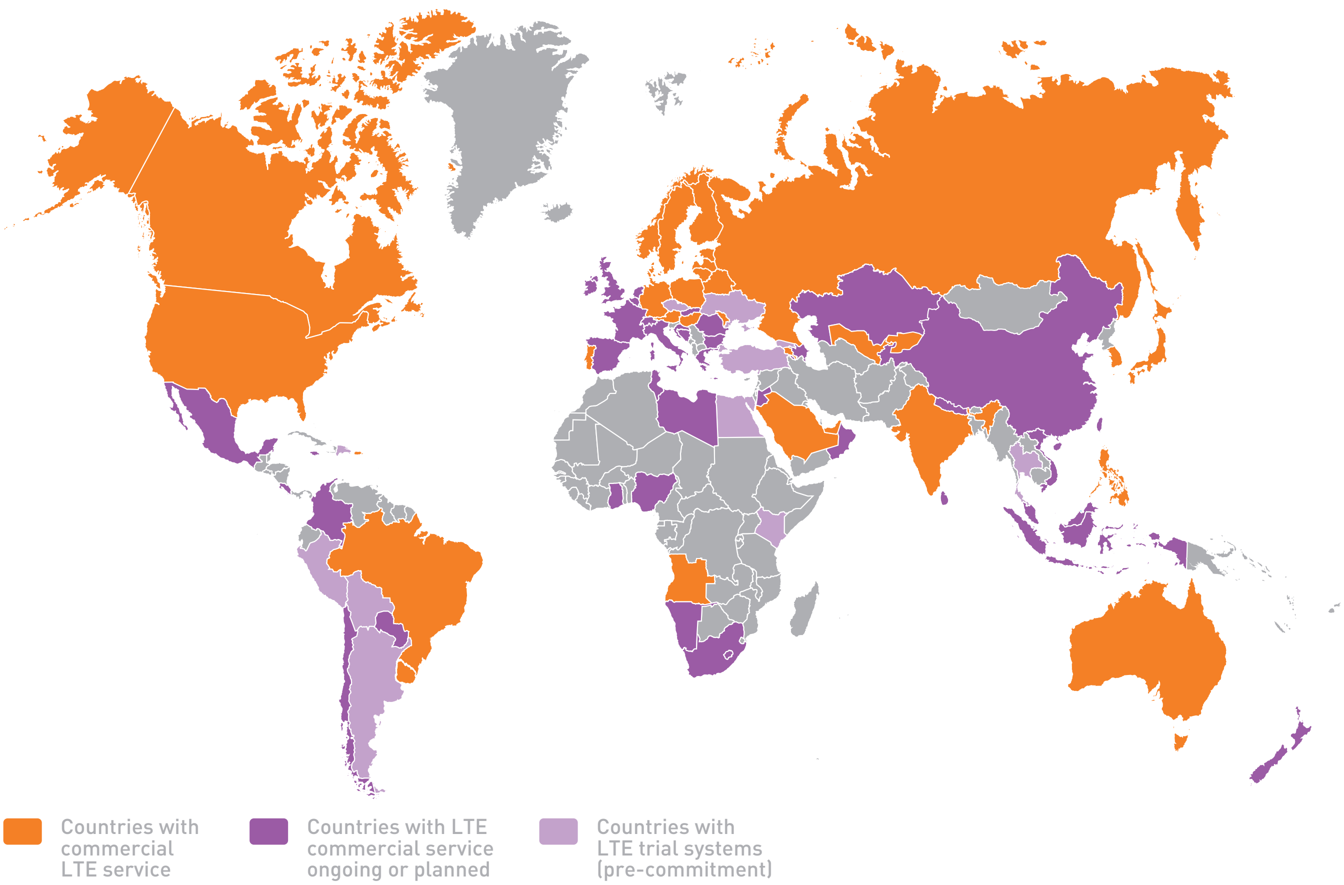
The 3GPP Long Term Evolution (LTE) high-speed, high-capacity data standard for mobile devices is well on its way to becoming a globally deployed standard. The first fully commercial LTE network was deployed in December 2009. In 2010, the first generation of CS telephony services via CDMA in combination with LTE PS data services were introduced, using dual always-on radios in users' mobile devices.

In 2011, LTE handsets using circuit-switched fallback (CSFB) to support GSM/WCDMA legacy systems became available, using smaller, less expensive, more power-efficient single radio solutions.

As of mid-2012, commercial LTE services in operation, planning or trials have expanded dramatically, as shown in Figure 1.

The availability of LTE-compatible smartphones early in the development of LTE networks has been a major driver of the rapid growth of LTE adoption. The prevalance of smartphones is driving the need for increased capacity, while users continue to demand faster speeds of access for their increasingly data-intensive applications. It is clear that the majority of operators, both GSM and CDMA, will evolve their networks to LTE.

Figure 1
May 2012 LTE Deployment and Plans



Source: Global Mobile Suppliers Association May 8, 2012 Evolution to LTE Report

The 3 phases of LTE voice and communication services evolution

The handling of voice traffic on LTE handsets is evolving as the mobile industry infrastructure evolves toward higher—eventually ubiquitous—LTE availability. This voice evolution can be characterized into three major phases, summarized in Figure 2.

In the first phase, currently under way, all voice traffic is handled by legacy Circuit-Switched (CS) networks, while data traffic is handled by LTE Packet-Switched (PS) networks—when and where available—and by legacy networks in non-LTE areas.

Despite the fact that LTE was originally designed as a data network, its quality of service (QoS) and capacity gains also provide subscribers and operators with significant additional benefits for voice services, such as HD voice, enhanced video capabilities, and rich communication offerings. Central to the enablement of LTE smartphones is meeting today's high expectation for mobile user experiences and evolving the entire communication experience by augmenting voice with these richer media services.

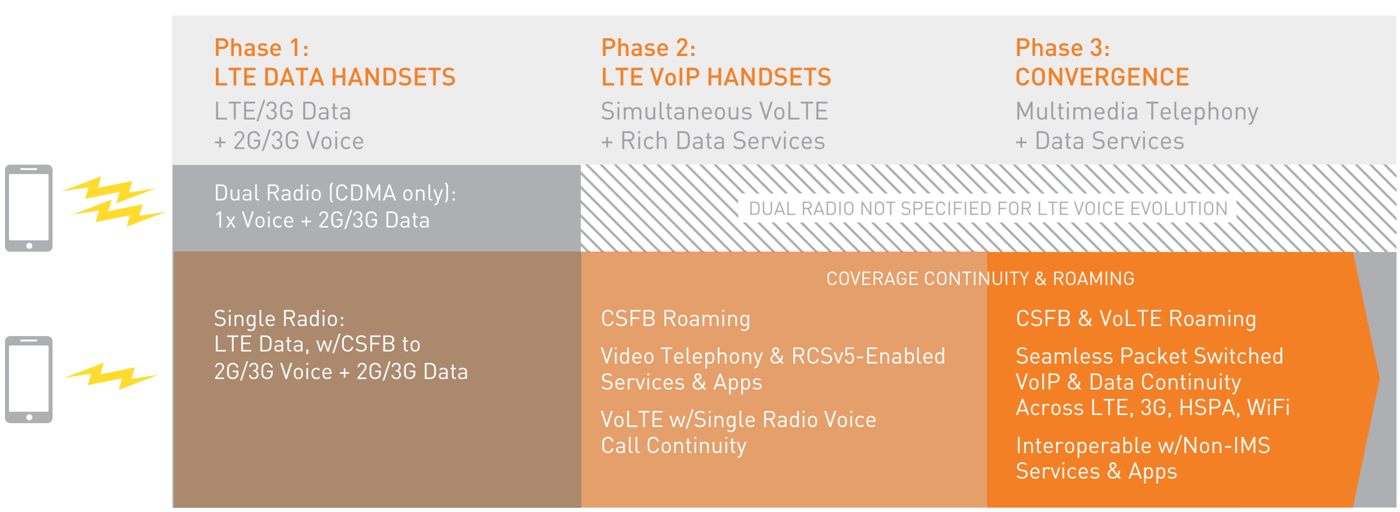
So the second natural step in LTE voice evolution is the introduction of voice over IP (VoIP) over LTE (VoLTE), a voice telephony solution comprising the IP Multimedia Subsystem (IMS) and the multimedia telephony (MMTel) service *(GSMA IR.92 [1])* that delivers voice services over LTE access. Based on IMS/MMTel, voice services can be further enriched with video *(GSMA IR.94 [2])* and combined with several other enhanced IP-based services such as HD voice, presence, location and Rich Communication Suite (RCS) additions like instant messaging, video share and enhanced/shared phonebooks.

This phase also uses a single radio solution in the user's device—with cost, size and battery efficiency advantages over dual radio solutions—with Single Radio Voice Call Continuity (SRVCC) that seamlessly maintains voice calls as mobile users move from LTE to non-LTE coverage areas. Without SRVCC, a VoLTE call on a device moving out of LTE coverage will be dropped, since no operators currently support VoIP on 3G.

The third phase of LTE voice evolution—all-IP network—converges the native power of IP to deliver enhanced capacity, value-added services (e.g., voice and video over IP and rich communication services) and interoperability across network access methods and operators (LTE, 3G/HSPA, WiFi and legacy telephony domains).

Figure 2

The 3 phases of LTE voice evolution



The benefits of VoLTE

Mobile VoIP using Internet-based PS 2G/3G applications (e.g. mobile Skype) have been available since early 2010. The data streams in these "over the top" (OTT) voice applications are not differentiated from other IP data traffic, so network and user device IP traffic loads can severely compromise voice quality of service.

VoLTE, in contrast, operates as a native application in the user's device, enabling prioritization over other data streams to deliver quality of service levels consistent with established user expectations.

Native VoLTE also provides enhanced native voice experiences not available with OTT VoIP, many of which users have become accustomed to in their 2G/3G telephony experiences, including:

- Wideband codecs, dual microphone, near/far noise cancellation
- Call continuity across networks, push-to-talk over cellular, group calling
- Video telephony
- Enhanced address book features such as presence/status, location, communication capabilities, pictures and other content syncing
- Rich media file sharing

For network operators, native VoLTE offers marketing and operating advantages, including:

- Enhanced user experiences and competitive offerings, including ARPU enhancement opportunities
- Greatly increased—and more efficient use of—network capacity, including use of new, wider and unpaired spectrum, fragmented spectrum and spectrum best suited to 10 MHz and above
- In areas where operators have LTE 700-900MHz licenses but limited or no 2G/3G voice coverage, VoLTE enables in-network voice service

The benefits of SRVCC

As an all-IP transport technology using packet switching, LTE introduces challenges to satisfying established quality of service expectations for circuit-switched mobile telephony and SMS for LTE-capable smartphones while being served on the LTE network.

In many cases, operators build and expand their LTE networks gradually, adding cells and capacity in line with their business plans and subscriber demand. As a result, LTE networks and the VoLTE services built on top of them must be able to coexist with legacy CS networks and to ensure handover to the legacy CS network when LTE coverage is insufficient. Since LTE and VoLTE services are a fundamental part of next-generation mobile networks, voice handover to legacy CS systems is a key capability while LTE coverage continues to be spotty.

The central challenge during the transition from today's hybrid networks with gaps in LTE coverage—both within single operator networks and across roaming agreement multi-operator networks—to an all-LTE network environment, and consequently all-VoLTE voice call environment, is transferring voice calls already in progress between LTE packet switched VOIP to legacy 2G/3G circuit switched voice, without compromising these established quality of service levels available in legacy networks today.

As noted before, in the first phase of voice evolution, all native voice calls are handled in the CS network. The LTE (PS data) connection falls back to the legacy 2G/3G CS voice network connection prior to initiation of a voice call.

In sharp contrast, a VoLTE call must be transferred from the LTE PS network to the legacy CS voice network *while the call is in progress*, while satisfying existing user experience expectations for minimal call interruption and low call drop rates. This handover needs to be well engineered with performance levels comparable to the Inter-Radio Access Technology (IRAT) handover procedures for voice calls available in CS networks today. These established QoS standards are less than 0.3 seconds voice interruption time and call drop rates under one percent.

SRVCC—Single Radio Voice Call Continuity—is the solution to this requirement for voice call continuity, and uses a single radio in the user's device along with upgrades to the supporting network infrastructure. This solution transfers VoLTE calls in progress from LTE to legacy voice networks, while meeting the rigorous QoS standards. Additionally, SRVCC—by ensuring voice call continuity—satisfies critical requirements for emergency calls.

Without SRVCC, operators with gaps or weaknesses in LTE coverage (or offering roaming in non-LTE networks) cannot realize the user experience and network efficiency advantages offered by VoLTE until LTE coverage is built out to match the full geographic range of their subscriber service commitments.

With SRVCC, operators can accelerate time to market and realize these benefits during the entire time span from today's hybrid network environments to the all-LTE environment of the future.

SRVCC network architecture

Since its initial specification in 3GPP Release 8, SRVCC has evolved continuously. To ensure interoperability of the various implementations with legacy networks, the GSMA has provided a set of guidelines for SRVCC *(GSMA IR.64 [4]),* detailing the requirements for networks and user devices. SRVCC provides continuity for PS to CS handover between LTE and WCDMA/GSM networks and from LTE to CDMA networks *(GSMA TS.23.216 [3]).*

GSMA guidelines recommend 3GPP Release 10 architecture for SRVCC (shown in Figure 3), because it reduces both voice interruption delay during handover and the dropped call rate compared with earlier configurations.

The network controls and guides the user device from LTE to 2G/3G as the user moves out of LTE coverage. The SRVCC handover mechanism is fully network controlled and calls remain under control of the IMS core network, which maintains access to subscribed services implemented in the IMS/MMTel service engine before, during and after the handover.

The Release 10 configuration includes all components needed to manage the time-critical signaling between the user's device and the network, and between network elements within the serving network, including visited networks during roaming. As a result, signaling follows the shortest possible path and is as robust as possible, minimizing voice interruption time caused by switching from the PS core to the CS core, whether the user's device is in its home network or roaming. With the industry aligned around the 3GPP standard and GSMA recommendations, SRVCC-enabled user devices and networks are interoperable, ensuring that solutions work well in all important call cases.

Network upgrades

As with any new handover technologies, SRVCC requires additional functionality in both the source (LTE) system and the target (legacy) system.

As specified in *GSMA IR.64 [4]* and summarized in Figure 4, SRVCC functionality can be added to the network by software updates of the MSS subsystem, the IMS subsystem and the LTE/EPC subsystems. No upgrades

Figure 3

SRVCC 3GPP R10 network architecture

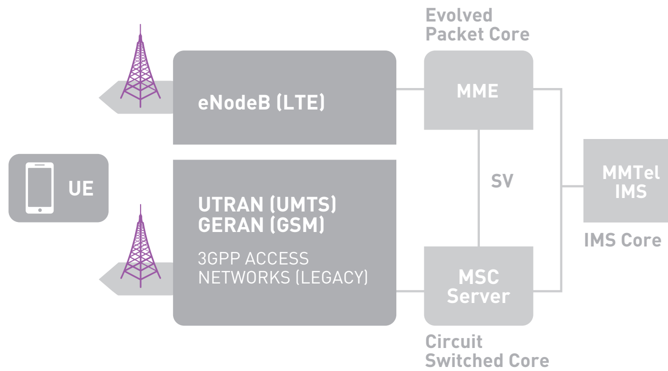
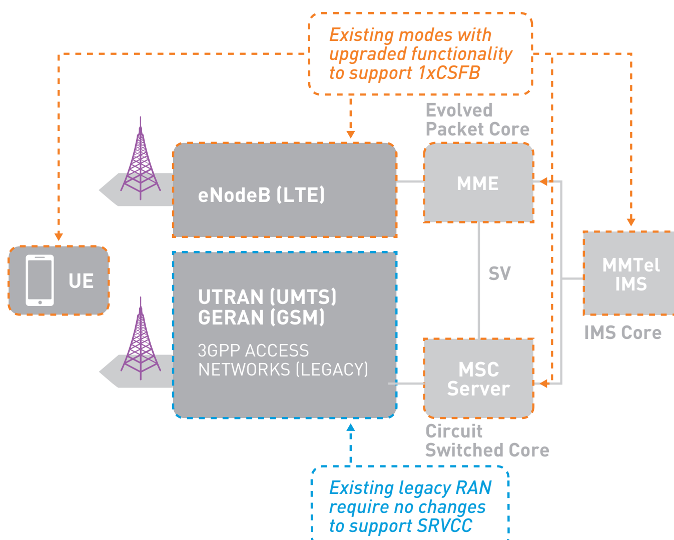


Figure 4

Software upgrades to support VoLTE



are required to the legacy GSM/WCDMA RAN target radio access.

Only a fraction of the installed base of Mobile Switching Center Servers (MSC-S) requires an upgrade to support well-functioning SRVCC. Where these servers are clustered in pools—the recommended architecture for the CS core—only the pools in the vicinity of the LTE coverage area need to be upgraded to achieve good performance. Additionally, only two servers in each pool need to be upgraded, with the second server upgraded primarily for backup redundancy.

Where it is not possible to upgrade a deployed MSC-S, either inside or outside a server pool, a dedicated MSC-S can be added to handle the SRVCC function with minimal impact on performance.

SRVCC voice handover process

When a user on an active voice call using VoLTE (Figure 5) moves into an area without LTE coverage, the handover of the call to the CS network is accomplished with two steps: *IRAT handover and session transfer*. IRAT handover is the traditional handover of the user's device from LTE radio access to WCDMA/GSM radio access. Session transfer is a new mechanism to move access control and voice media anchoring from the LTE Evolved Packet Core (EPC) to the legacy CS core.

During the entire voice handover process from LTE to 2G/3G, the IMS/MMTel retains control of the user.

The handover process is initiated by a session transfer request (Figure 6) to the IMS/MMTel.

The IMS/MMTel responds simultaneously with two commands, (Figure 7) one to the LTE network and one to the 2G/3G network. The LTE network—on which the user's voice call is in progress—receives an IRAT handover execution command through the MME and the LTE RAN to the instruct the user's device to prepare to move to the CS network for the voice call. The CS network—where the user's voice call is being sent—receives a session transfer response, preparing it to accept the call in progress.

Figure 5
VoLTE call in progress

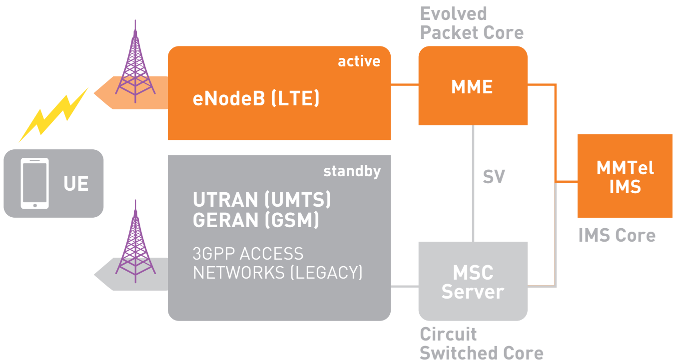


Figure 6
Session transfer request

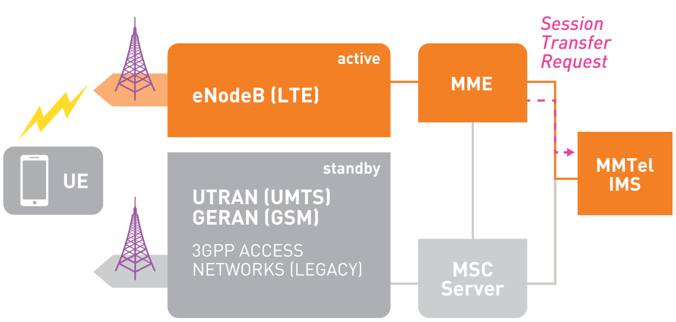
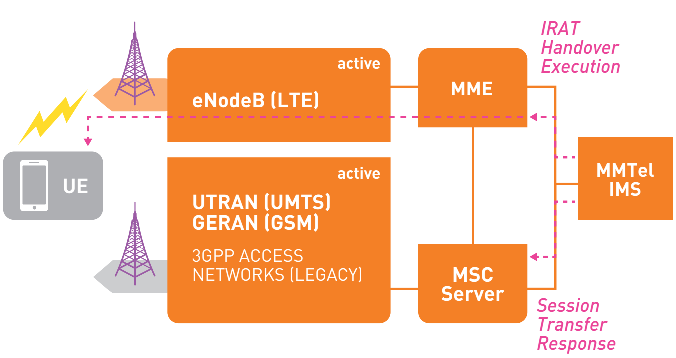


Figure 7
Simultaneous IRAT handover and session transfer commands



With acknowledgements that the commands have been executed, the user's device and the IMS/MMTel—still in control of the user's voice call in progress—switch to the CS network to continue the call. (Figure 8)

Return to LTE on call completion

While the implementation of SRVCC does not directly impact the legacy GSM/WCDMA RAN nodes, an indirect impact arises from methods to help the user device return to LTE PS data access as soon as possible after the voice call has ended. To help guide the device back to LTE, the legacy RAN can implement functionality to either:

- Broadcast LTE system information to the user's device, with which the device can perform a cell reselection an LTE cell after connection release (if the legacy RAN is configured to do this)
- Release connection to the user's device and simultaneously redirect it to connect to LTE

Voice interruption time

The two parallel procedures (IRAT handover and session transfer), as well as the number of subsystems needing updates, have raised concern whether SRVCC can meet the 3GPP *(TS22.278 [5])* voice interruption performance target of less than 0.3 seconds. Both the IRAT handover in the RAN and the session transfer in the core network contribute to voice interruption time as they break and remake the connection.

To address this interruption performance target, starting at the design level, the SRVCC 3GPP R10 network architecture (Figure 3) minimizes the interruption by initiating the two procedures simultaneously so they can run in parallel.

Performance testing has revealed that the session transfer process is the quicker of the two parallel procedures, on the order of 0.01 second. Such quick media redirection, moving the media anchor from the EPS domain to the MSS domain, means that voice interruption time is influenced primarily by IRAT handover delay.

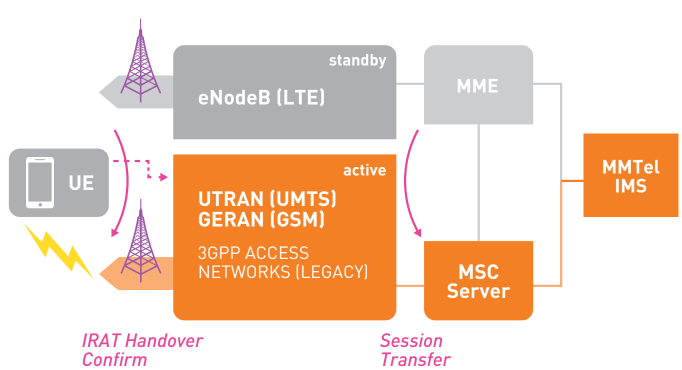
IRAT handover delay is the time from the user device's receipt of the network handover command (Figure 7) to the time it is synchronized on the new radio access and has sent a confirmation message (Figure 8). Total voice interruption time is slightly greater than IRAT handover time alone, and is defined by the time from when the last voice packet is sent over LTE until voice media is sent over CS access.

Testing using commercially available network infrastructure (Ericsson) and commercially available chipset-based test phones (Qualcomm) has shown voice interruption time on par with legacy CS IRAT handovers between WCDMA and GSM, and inter-frequency handovers within a RAT, when the RAN orders the user device to retune the radio to a new frequency. This measured voice interruption time is within the 3GPP target of less than 0.3 seconds.

A related performance metric is the time it takes for the network to prepare the SRVCC handover. This is the time between when the user device measures and reports inadequate LTE reception and when the user device receives the handover command and executes the handover. While handover preparation time does not interrupt the user's voice call, it needs to be short to avoid the risk that LTE reception deteriorates between initiation and handover and the subsequent handover fails for lack of adequate LTE availability.

Testing with commercially available network infrastructure (Ericsson) and test phones using commercially available chipsets (Qualcomm) has shown that handover preparation and procedure time is on par with legacy CS IRAT handovers between WCDMA and GSM, and inter-frequency handovers within a RAT, when the RAN orders the user device to retune the radio to a new frequency, averaging about 0.2 seconds for handover.

Figure 8
Voice transfer to CS



Call retention

Since the aim of SRVCC is to greatly reduce the number of dropped voice calls caused by users moving in and out of LTE coverage, the SRVCC mechanism must work not only for ordinary voice calls—the bulk of the traffic—but also for the small yet vital volumes of emergency calls. It must support the deployment of an LTE network in line with network operators' business strategies and subscriber demand, and should support voice handover to and from VoLTE and CS telephony over 2G/3G access.

To meet these requirements, the SRVCC specification is designed to provide good call retention in the following scenarios:

- During the active phase of a call, including IMS emergency calls
- During the call initiation alert phase, with handover executed when a user initiates an outgoing call or receives an incoming call
- During an inactive (on hold) call, both with and without an active call
- While participating in a conference call
- While on an IMS-based video call, maintaining voice stream continuity while (under current specifications) allowing video streaming to end during SRVCC handover

The call retention rate is the percentage of successfully completed calls for a given user. Since SRVCC can happen only once per call, and some calls may include both SRVCC handover and other handovers, the SRVCC handover success must be higher than the total call retention target rate. With legacy voice call retention rates typically higher than 98%, SRVCC handover is targeted to be successful more than 99% of the time.

Accurate evaluation of call retention requires a statistically significant amount of data over a variety of different real-world radio conditions. Such volumes of data for SRVCC handover are, expectably, not yet available. An indication of call retention probability can, however, be obtained by examining similar systems. As with service interruption time, call retention probability is dependent on two factors:

- The probability of IRAT handover failure, and
- The probability of session transfer failure

IRAT handover failure occurs more frequently in real-world deployments than session transfer failure, so this measurement is a better predictor of call retention probability. From a reliability perspective, SRVCC handover to WCDMA is similar to WCDMA inter-frequency handover (IFHO). The time to do the handover preparation of the target WCDMA cell is also the same. So call statistics for IFHO may provide some insight into the expected call retention rate for SRVCC.

Based on data collected from existing commercial deployments, the percentage of successful handovers for IFHO in well-planned networks is in the 98% to 99% range. Most of the calls for which handover fails return successfully to the original cell and continue, so the actual failure rate is typically significantly less than 0.5%. Given its similarity to IFHO, similar statistics are likely for SRVCC. Should call retention become a problem in certain network coverage areas, call retention might be improved by tuning handover parameter settings.

Additionally, as an early indicator, testing to date using Ericsson commercial products and Qualcomm test phones with commercial chipsets has already demonstrated SRVCC handover success rates greater than 99%.

Conclusions

VoLTE offers numerous user experience and operator network benefits in the rapidly growing LTE network environments, unachievable in any other way.

SRVCC is a key functionality in the implementation of a stepwise LTE and VoLTE deployment, interoperable with legacy networks. To ensure success, the GSMA has aligned the industry to ensure VoLTE and SRVCC deployments follow a set of recommendations that secure interoperable implementations for both a networks and user devices.

Interoperability development and performance testing has shown that VoLTE with SRVCC is operational and capable of meeting the quality of service performance specifications for voice interruption and call retention.

The impact on legacy networks is manageable; while software upgrades are required for the LTE/EPC/IMS subsystems, there is limited—or no—impact on the legacy RAN, and only a fraction of deployed MSC-Ss need to be upgraded.

Consequently, the second phase of LTE voice evolution—VoLTE with SRVCC in hybrid LTE/2G/3G network environments—is now ready to move toward operational deployment.

This white paper has been developed in collaboration with Ericsson.

References

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Abbreviations used

2G	2nd-generation wireless telephone technology
3G	3rd-generation wireless telephone technology
3GPP	3rd Generation Partnership Project
CDMA	code division multiple access
CS	circuit-switched
CSFB	circuit-switched fallback
EPC	Evolved Packet Core
FDD	frequency division multiplexing
GSM	Global System for Mobile Communications
GSMA	GSM Association
HD	high definition
IFHO	inter-frequency handover
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IRAT	Inter-Radio Access Technology
LTE	Long Term Evolution
LTE-FDD	LTE network using FDD
LTE-TDD	LTE network using TDD
MME	Mobility Management Entity
MMTel	multimedia telephony
MSC	mobile switching center
MSC-S	Mobile Switching Center Server
MSS	mobile softswitch
PS	packet switched
RAN	radio-access network
RAT	radio-access technology
RAU	radio-access unit
SRVCC	Single Radio Voice Call Continuity
TDD	time-division multiplexing
VCC	Voice Call Continuity
VoIP	voice over IP
VoLTE	voice over LTE
WCDMA	wideband CDMA