Telephony services over LTE end-to-end

How to deliver telephony services for LTE-capable devices is a topic of much debate. Ericsson supports two standardized solutions for delivering these services – IMS Multimedia Telephony and circuit-switched fallback for use by operators prior to migrating to IMS/MMTel.

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Although LTE is a packet-only technology, the standard was defined to efficiently handle voice and multimedia services, providing support for more than mobile internet access.

This article describes how Ericsson and telecommunications standardization bodies envision the realization of telephony services over LTE/EPC.

The authors take an end-to-end perspective, outlining requirements in the LTE radio access, EPC, IMS core, and MMTel application server. They describe IMS features for voice telephony, media handling, radio features for voice, QoS quality, roaming architecture, emergency calls, and solutions that coexist with circuit-switched infrastructure.

Introduction to telephony services over LTE

The 3rd Generation Partnership Project (3GPP) Long Term Evolution (LTE) standard promises to deliver efficient mobile broadband and multimedia communication services. In keeping with this objective, it was developed to include the features needed to effectively support voice-over-IP (VoIP) media and control. How telephony services are to be realized over LTE, however, has been hotly debated in recent years.

Ericsson stands behind the two solutions standardized by 3GPP, namely IP Multimedia Subsystem (IMS) Multimedia Telephony (MMTel) and circuit-switched fallback (CSFB). Ericsson is promoting MMtel as the main telephony-over-LTE solution because it fulfills all the requirements of a telephony solution. What is more, it is future-proof, allowing an evolution from today's voice and video telephony to full-fledged multimedia communication.

Fallback to second- and third-generation (2G/3G) circuit-switched telephony is promoted as a migration step when moving from circuit-switched networks to all-IP networks and MMTel. When a voice call is placed or received, the CSFB solution shifts the user equipment (UE) access from LTE to 2G/3G.

To align the industry in support of 3GPP standards Ericsson and several other vendors and network service providers jointly formed the One Voice initiative, whose main objective is to outline a baseline profile with which terminal and network vendors should comply to guarantee interoperability. The work of the initiative has gained wider acceptance in the industry since its adoption by the GSMA in the beginning of 2010.

The baseline profile produced by the One Voice initiative forms the basis of

BOX B

Circuit-switched fallback is an initial mechanism to allow an LTE-capable smartphone to move to the circuit-switched domain when handling voice calls.

BOX A Terms and abbreviations

3GPP	3rd Generation Partnership Project
AKA	authentication and key agreement
AMR	adaptive multi-rate
CSFB	circuit-switched fallback
DRX	discontinuous reception
eNB	eNode-B or base station
EPC	evolved packet core
EPS	3GPP Evolved Packet System
GBR	guaranteed bit rate
GSM	global system for mobile communications
GSMA	GSM Association
ICS	IMS centralized services
IMS	IP Multimedia Subsystem
IP	internet protocol
ISIM	IMS SIM
ITU	International Telecommunication Union
LTE	Long Term Evolution
MMTel	IMS Multimedia Telephony
MSISDN	mobile subscriber ISDN number
OTDOA	over-the-air time difference of arrival
PCC	policy and charging control
PCRF	policy and charging rules function
P-CSCF	proxy call session control function
PDCP	packet data convergence protocol
QCI	QoS class identifier
QoS	quality of service
RLC	Radio Link Control
ROHC	robust header compression
SIM	subscriber identity module
SIP	session initiation protocol
SMS	short message service
SR-VCC	single radio VCC
TTI	transmission time interval
UE	user equipment
URI	uniform resource identifier
USIM	universal SIM
VCC	voice call continuity
VoLTE	voice-over-LTE
VoIP	voice-over-IP
XCAP	XML configuration access protocol
XML	Extensible Markup Language

the *IMS profile for Voice and SMS* (GSMA permanent reference document IR.92), commonly referred to as voice-over-LTE (VoLTE).

The requirements in the baseline profile span every layer of the network. Accordingly, they comprise IMS features, media requirements, bearer management features, LTE radio requirements, and common functions, such as the IP version (Figure 1). The requirements are compliant with 3GPP Release 8 (Rel-8) and there are some additional requirements from 3GPP Release 9 (Rel-9) in support of packet-switched emergency calls.

IMS features for voice

The VoLTE profile uses a subset of general IMS features and of the MMTel service. The subset is selected to provide an IP telephony service with a similar user experience as the circuit-switched service used today in GSM and 3G networks.

The aim of the profile is to provide a minimal set of features that can serve as a starting point. The profile must still be open to functional growth in networks and devices to reach the full 3GPP specifications for IMS and MMTel, for example, using multimedia, supporting users with multiple devices or the full set of MMTel supplementary services.

Considerable efforts have therefore been made to ensure that user equipment (UE) that supports the VoLTE feature set can be used even in networks that provide all 3GPP IMS and MMTel features. Likewise, efforts have been made to guarantee that a network that only supports the VoLTE feature set can offer this limited service to users of UEs in which the richer 3GPP IMS/MMTel service set has been implemented (Figure 2).

The baseline profile mandates the use of different options for the IMS features – for example,

- SIP registration;
- authentication;
- addressing;
- basic call handling;
- forking; and
- signaling compression.

Where authentication is concerned, the baseline profile recognizes that the target device is a phone that uses cellular access and mandates support for IMS authentication and key agreement (IMS-AKA) and IMS subscriber identity module (ISIM). A universal SIM (USIM) may be used if the network service provider has not deployed ISIM.

The baseline profile supports both mobile subscriber ISDN number (MSISDN) and alphanumeric session initiation protocol-uniform resource identifier (SIP-URI)-based addressing. The terminals are forward-compatible and may be used in scenarios where network service providers employ *name@operator.com* types of addressing.

The baseline profile supports a single handheld device scenario. Forking is thus not required in the network. To be forward compatible, however, the UE must be usable in a network where multiple devices per user are supported in IMS.

Signaling compression can be useful for minimizing the SIP signaling load over the air interface. Support for this option is mandated in the UE as well as the network but network service providers can turn it on or off.

The set of supplementary services in VoLTE was chosen to provide service parity with existing GSM and 3G services. The MMTel specification includes more features – for example, a more flexible Extensible Markup Language (XML) configuration access protocol

BOX C

IMS Centralized Services is a key enabler for IMS/MMTel, allowing call control for a mobile device to be anchored in the IMS domain through the circuit-switched domain when not in LTE coverage.

FIGURE 1 **The VoLTE specifications.**

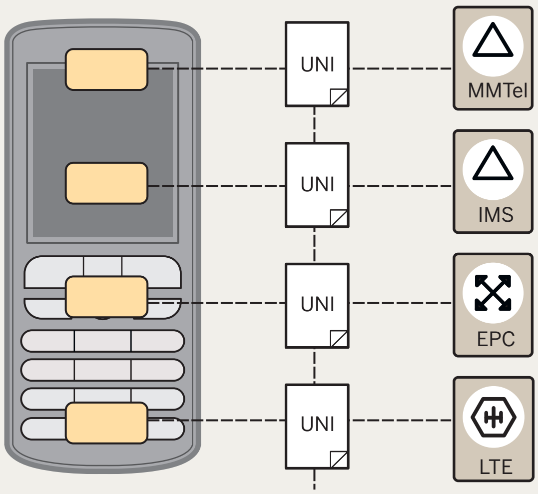


FIGURE 2 **Upwards compatibility of the VoLTE profile.**

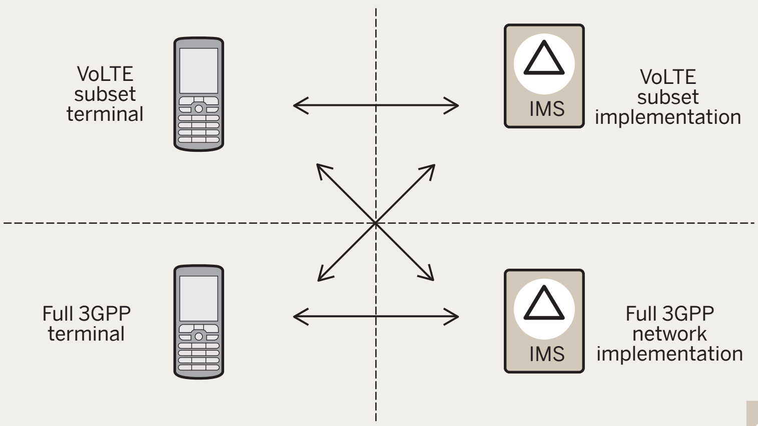
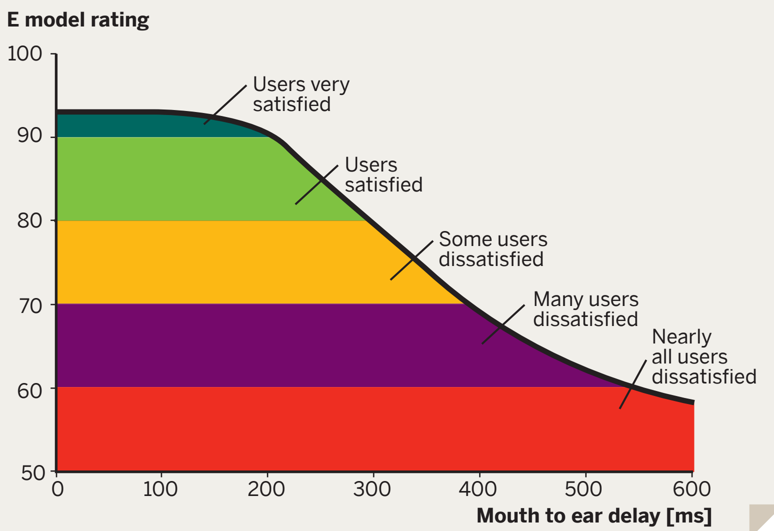


FIGURE 3 **User satisfaction versus end-to-end (mouth-to-ear) delay.**



BOX D

Single radio voice call continuity (SRVCC) allows a mobile with an ongoing voice call to transition to the circuit-switch domain in the event of loss of LTE coverage.

(XCAP)-based service-management capability that allows black- or white-list-based and time-dependent diversions. While the VoLTE profile retains this basic mechanism for upward compatibility, the minimum defined baseline profile uses only basic GSM-like controls.

Media handling and control

The objective for media handling in the VoLTE baseline profile was to have a set of requirements that gives voice service with conversational quality on par with that of legacy circuit-switched service.

To ensure interoperability, in line with 3GPP, the baseline profile mandates support of the adaptive multi-rate (AMR) narrowband speech codec. However, seen in terms of an MMTel business solution, the recommended speech codec for voice over LTE access is AMR wideband (high-definition voice). This codec is mandated in 3GPP networks and terminals that support 16KHz speech sampling.

For a given codec and the set of front-end handling requirements, two variables influence speech quality the most: end-to-end delay and frame-error rate. Figure 3 shows the relationship between user satisfaction and end-to-end delay according to the International Telecommunication Union (ITU) E-model. User satisfaction with communication quality is greatest when end-to-end delay is under 200ms, which is the typical end-to-end delay of 2G/3G circuit-switched systems.

From an end-to-end latency point of view, the low latency of LTE access (20-30ms round-trip delay) means the minimum end-to-end delay may be significantly lower than 200ms. However, LTE radio bearers do not employ fixed delay. Instead, they use fast retransmissions to repair erroneous transmissions, and LTE uses schedulers to control uplink and downlink transmissions. Consequently, LTE transmissions introduce delay variations (jitter), and the endpoints that terminate the voice flow must implement efficient de-jitter buffers.

Notwithstanding, despite the additional delay introduced by de-jittering, the end-to-end delay of commercial MMTel terminals and LTE networks is on par with or better than that of 2G/3G circuit-switched telephony.

Where frame error rates are concerned, LTE voice bearers add erroneous transmission repair (2G/3G circuit-switched voice bearers do not). As a result, most packets are sent successfully over the LTE air interface, and the residual frame error rate for voice media is low.

The addition of solutions for codec rate adaptation, which is specified in the MMTel media specification, can further enhance media transport. In high load situations, for example, codec rate adaptation can be used to increase system capacity and coverage by lowering the codec rate.

Quality of service (QoS) control

The service requirements put on bearer management and charging mandate that an LTE/evolved packet core (EPC) system, which provides access for MMTel, must be equipped with policy and charging control (PCC).

PCC introduces a set of *Rx* and *Gx* interfaces that connect EPC with the IMS domain. In addition, the PCC architecture introduces a policy and charging rules function (PCRF) that encompasses policy decision and flow-based charging control functions.

With PCC implemented, IMS can start a network-initiated process to set up suitable 3GPP Evolved Packet System (EPS) bearers for telephony service. The characteristics of these bearers are signaled via a QoS class identifier (QCI). Two standardized classes are

- **QCI1 – a guaranteed bit-rate bearer for VoIP media; and**
- **QCI5 – a high-priority non-guaranteed bit-rate bearer for IMS control messages that use the session initiation protocol (SIP) and the XCAP.**

The LTE RAN, in other words, the e-Node-B or base station (eNB) is responsible for admission control for the EPS bearers. To determine if a new bearer can be admitted, the eNB must ensure that there is sufficient capacity across the air interface, the transport network, and on internal software or hardware resources. Once admitted, the eNB will use the EPS QoS parameters to invoke the appropriate eNB scheduling policies to meet the required end-to-end delay and packet loss rate targets.

PCC helps control charging for the service. Charging for the telephony session is managed in the IMS domain; therefore, charging on the EPS bearer level is set to zero. The PCRF is used to set the proper bearer-charging rules for EPS bearers.

An important role of PCC is to handle error cases when the EPS bearer for voice service is lost – for example, when a UE travels outside radio coverage. Using guaranteed bit-rate (GBR) bearers LTE/EPC notifies the PCRF when the EPS bearer for voice is lost. The PCRF then informs the proxy call session control function (P-CSCF) in the IMS domain, which terminates the session and charging.

Radio features and performance for voice

Ordinarily, the radio network manages the most constrained resource, and as a result, it must balance a number of competing objectives, such as coverage, capacity, quality and battery longevity (talk time).

To maximize coverage, which is typically uplink-limited, the eNB primarily employs features to reduce the packet size. For this purpose, it employs robust header compression (ROHC) and radio link control (RLC) unacknowledged mode with short packet data convergence protocol (PDCP)/RLC sequence numbers. By reducing the packet size, greater coding redundancy is obtained for a given uplink allocation, thereby improving the reliability of data delivery in coverage limited scenarios. The eNB can also employ features such as

- **transmission time interval (TTI) bundling to allow a UE to concentrate energy on narrow uplink allocation for a longer duration; and**
- **frequency hopping to reduce the effects of interference across multiple cells.**

To maximize capacity, in other words, to maximize the number of UEs that can transmit or receive VoIP packets in the cell, the eNB leverages the same tools to reduce the packet size as mentioned above. Further enhancements can be achieved by adding Explicit Congestion Notification (ECN) to reduce the speech codec rate between the two endpoints. For some UEs, bundling of VoIP packets can help reduce packing inefficiencies, as long as the delay targets can be met. From a control channel perspective, the eNB may need to employ techniques such as predictive grant allocation to reduce the amount of signaling required to control a large number of UEs.

To maintain quality, the eNB must ensure that end-to-end delay budgets and packet-loss rates are met for media packets. For packet-delay budgets, to achieve end-to-end (mouth-to-ear) delays of 200ms, network service providers typically configure the eNB for targets in the range of 50-80ms. The challenge is that the eNB must schedule a large number of UEs in a limited set of scheduling resources, taking into account the need to retransmit packets until successfully received. The packet loss rate is typically controlled in the link-adaptation algorithms to statistically ensure that the VoIP packets are successfully transmitted within the loss targets. The packet loss rate and the packet delay budget must each be accounted for in the mechanisms that are used for cell handover.

Finally, a very important aspect of VoIP support on the radio access network is the need to maximize user talk time. LTE allows UEs to turn off their transceiver periodically between VoIP frames. A combination of appropriately configured active discontinuous reception (DRX) and semi-persistent scheduling can potentially offer significant improvements over today's solutions.

Roaming architecture

Service control in IMS is home-network-based. This allows network service providers to differentiate services, but might make it more complex to deliver the service to roaming users. To enable a split of service revenue and to meet regulatory requirements, the visited network needs to be aware of telephony services even when the service is provided from the home network.

To support this, the VoLTE profile uses a roaming model where the IMS P-CSCF is located in the visited network and uses local breakout from the packet core network **(Figure 4)**. IMS voice services for roaming users therefore require new IMS-based roaming agreements between the network service providers. Until such agreements are in place, the UE is required to use CSFB for voice calls while roaming if LTE is used in the visited network.

Emergency calls

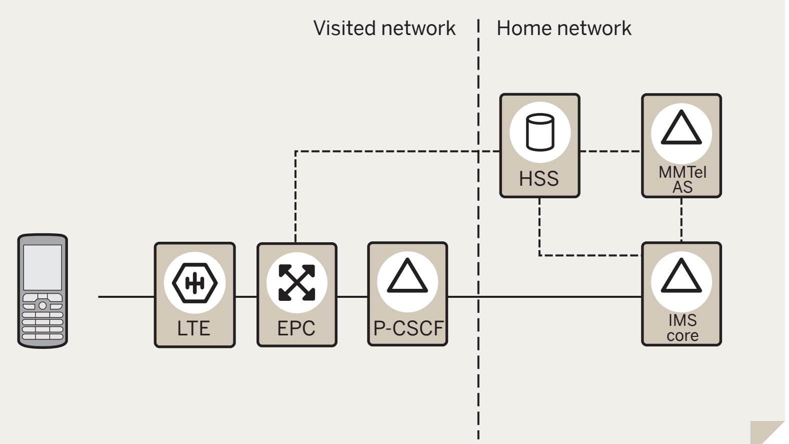
Some voice-over-LTE deployments can rely on continuous circuit-switched network coverage for emergency calls. But for deployments that only use VoIP technology or where the LTE coverage in some locations is better than the circuit-switched coverage, emergency calls must be able to use IMS. Therefore, the VoLTE profile includes baseline specification for IMS emergency calls.

3GPP Rel-9 added the functions in LTE/EPC that enable IMS-based emergency calls over packet-switched radio access. From an LTE/EPC perspective, 3GPP Rel-9 includes support for SIM-less emergency calls, user- and control-plane location services, and enhancements to the air interface to improve over-the-air time difference of arrival (OTDOA) accuracy. These additions were made to support more stringent international regulatory requirements.

Legacy coexistence

In many networks, the coverage of the voice-capable LTE deployments will be less extensive than that of circuit-switched voice. Initially, where LTE coverage is spotty, LTE UEs can use circuit-switched fallback procedures. This

FIGURE 4 IMS roaming architecture.



way, LTE users who receive or place a call temporarily (for the duration of the call) fall back to the circuit-switched network. CSFB is not part of the VoLTE profile as it does not require IMS voice.

Once VoLTE has been introduced, the handover of ongoing calls to the circuit-switched domain might still be needed to reduce the number of dropped calls. The VoLTE specification includes optional single radio voice call continuity (SR-VCC) procedures for handing over calls to the circuit-switched network. IMS centralized services (ICS) allow the IMS service domain to be used for IMS users when the access is provided via a circuit-switched network. These migration aspects are further described in *Achieving a converged service offering for fixed and mobile telephony.*[2]

Summary

The companies in the VoLTE initiative and the GSMA have agreed on how the telephony service is to be realized over LTE access. 3GPP MMTel specifications make up the basis of this realization. The result is a feature-rich telephony service whose performance is on par with GSM circuit-switched telephony.

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