1

LTE Basement

Mobile networks are rapidly transforming—traffic growth, bit rate increases for the user, increased bit rates per radio site, new delivery schemes (e.g., mobile TV, quadruple play, IMS), and a multiplicity of RANs (2G, 3G, HSPA, WiMAX, LTE)—are the main drivers of the mobile network evolution. The growth in mobile traffic is mainly driven by devices (e.g., smartphone and tablet) and applications (e.g., mainly web browsing and video streaming). To cope with the increasing demand, mobile networks have based their evolution on increasingly IP-centric solutions. This evolution relies primarily on the introduction of IP transport, and secondly, on a redesign of the core nodes to take advantage of the IP backbones.

The first commercial LTE network was opened by Teliasonera in Sweden in December 2009, and marks the new era of high-speed mobile communications. The incredible growth of LTE network launches boomed between 2012 and 2016 worldwide. It is expected that more than 500 operators in nearly 150 countries will soon be running a commercial LTE network. Mobile data traffic has grown rapidly during the last few years, driven by the new smartphones, large displays, higher data rates, and higher number of mobile broadband subscribers. It is expected that the mobile broadband (MBB) subscriber numbers will double by 2020, reaching over 7 billion subscribers, that MBB data traffic will grow fourfold by 2020, reaching over 19 petabytes/month. The internet traffic, MBB subscriber, and relative mobile data growth is illustrated in Figure 1.1.

1.1 LTE Principle

To provide a fully mature, real-time–enabled, and MBB network, structural changes are needed in the network. In 2005, the 3GPP LTE project was created to improve the Universal Mobile Telecommunications System (UMTS) mobile phone standard to cope with future requirements, which resulted in the newly evolved Release 8 (Rel 8) of the UMTS standard. The goals include improving efficiency, lowering costs, improving services, making use of new spectrum opportunities, and better integration with other open standards. Long-term evolution (LTE) is selected as the next generation broadband wireless technology for 3GPP and 3GPP2. The LTE standard supports both FDD (frequency division duplex), where the uplink and downlink channel are separated in frequency, and TDD (time division duplex), where uplink and downlink share the same frequency channel but are separated in time. After Rel 8, Rel 9 was a relatively small update on top of Rel 8, and Rel 10 provided a major step in terms of data rates and capacity with carrier aggregation, higher-order Multi-Input-Multi-Output (MIMO) up to eight antennas in downlink and four antennas in uplink. The support for heterogeneous network (HetNet) was included in Rel 10, also known as LTE-Advanced (Figure 1.2).

LTE Optimization Engineering Handbook, First Edition. Xincheng Zhang.

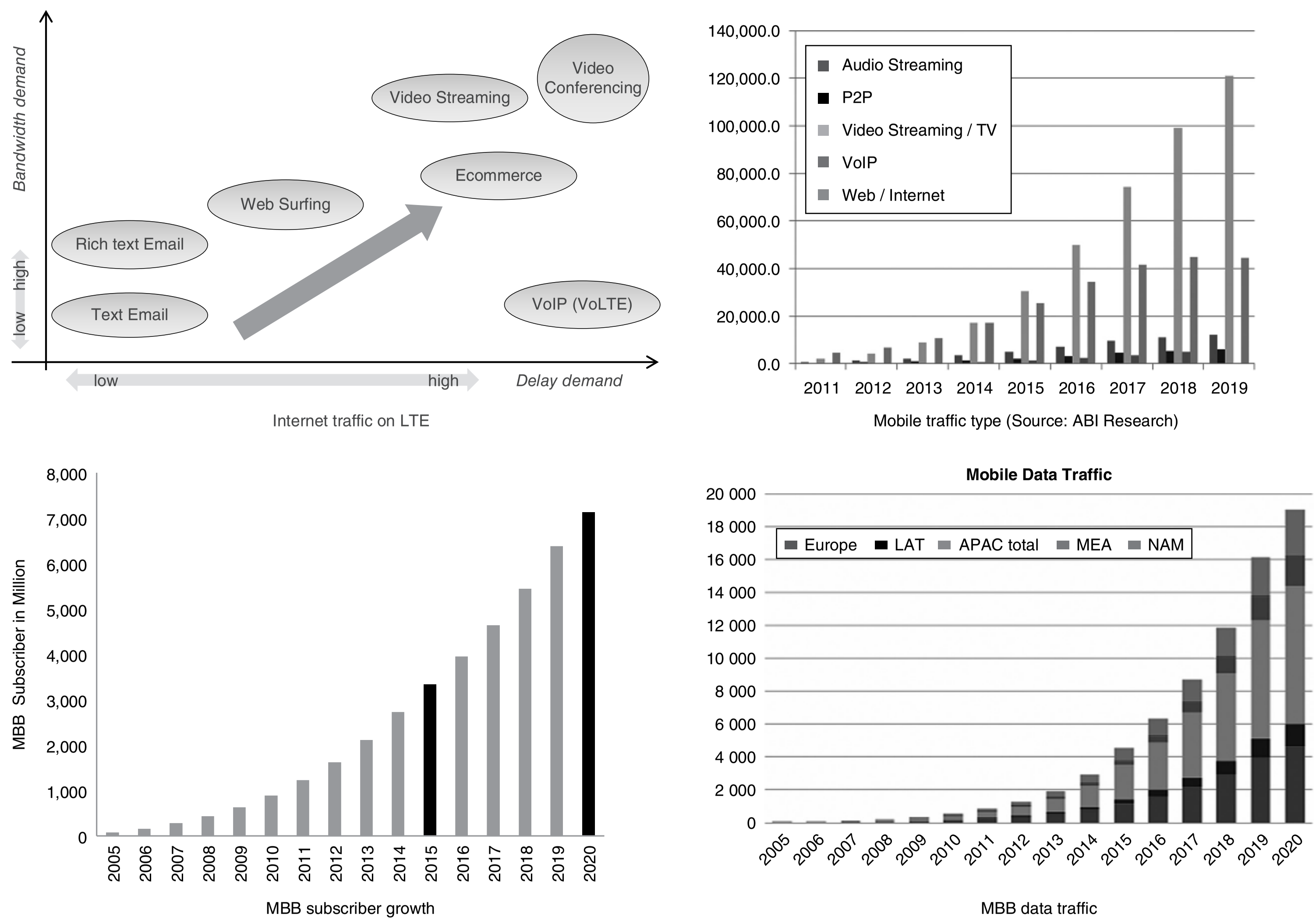


Figure 1.1 The internet traffic, MBB subscriber, and relative mobile data growth.

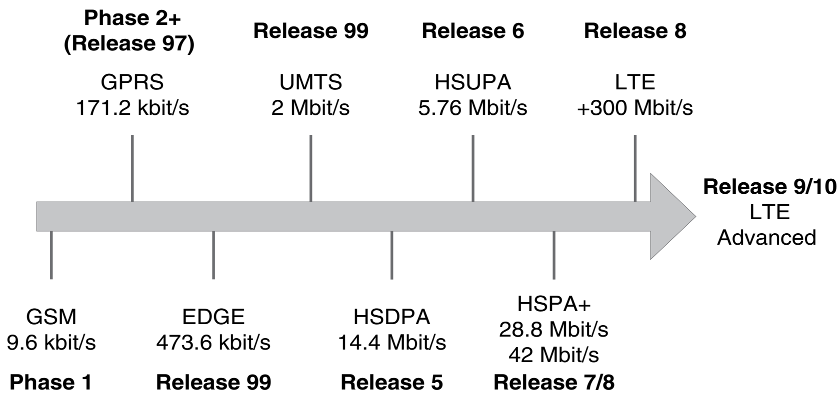


Figure 1.2 3GPP standard evolution.

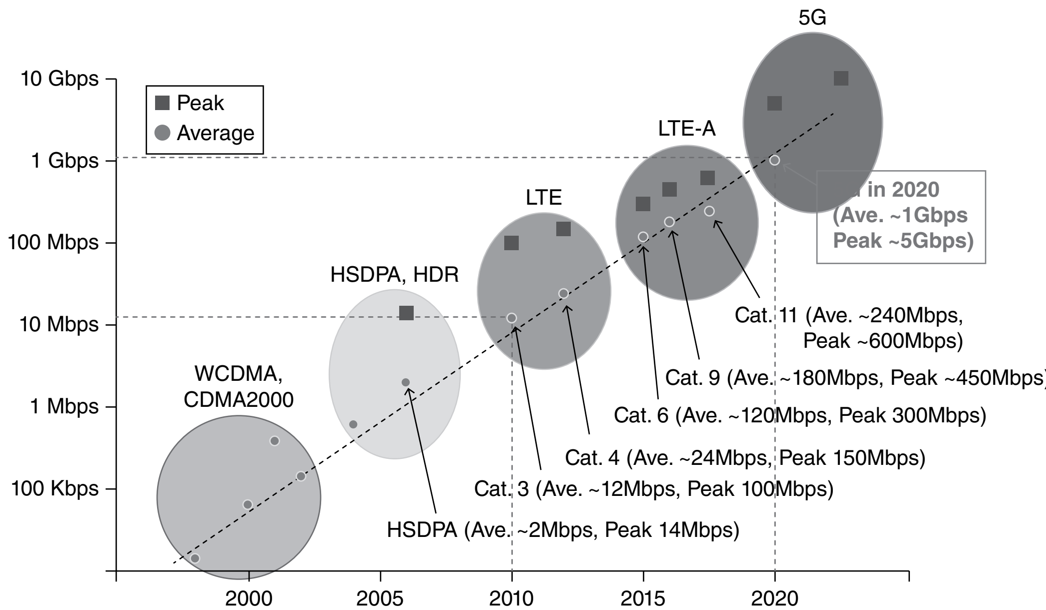


Figure 1.3 Downlink data rate evolution.

Among the design targets for the first release of the LTE standard are a downlink bit rate of 100 Mbit/s and a bit rate of 50 Mbit/s for the uplink with a 20-MHz spectrum allocation. Smaller spectrum allocation will of course lead to lower bit rates and the general bit rate can be expressed as 5 bits/s/Hz for the downlink and 2.5 bits/s/Hz for the uplink. Rel 10 (LTE-Advanced), was completed in June 2011 and the first commercial carrier aggregation network started in June 2013 (Figure 1.3).

LTE provides global mobility with a wide range of services that includes voice, data, and video in a mobile environment with lower deployment cost. The main benefits of LTE include (Figure 1.4):

- Wide spectrum and bandwidth range, increased spectral efficiency and support for higher user data rates

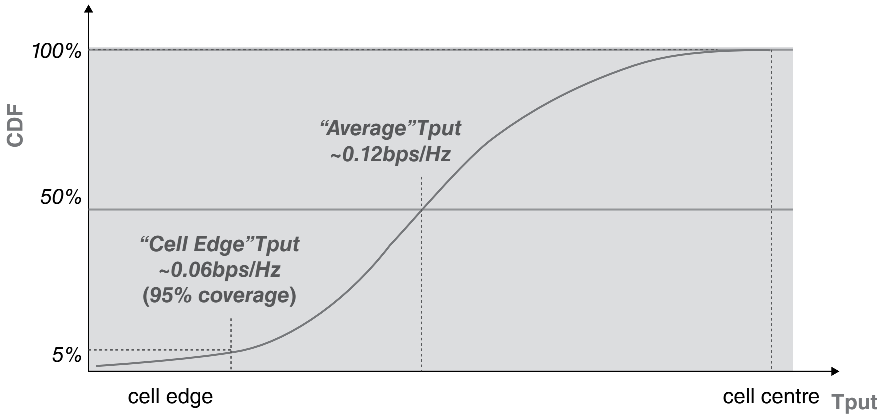


Figure 1.4 Throughput of a user, 10 users evenly distributed in cell.

- Reduced packet latency and rich multimedia user experience, excellent performance for outstanding quality of experience
- Improved system capacity and coverage as well as variable bandwidth operation
- Cost effective with a flat IP architecture and lower deployment cost
- Smooth interaction with legacy networks

LTE air interface uses orthogonal frequency division multiple access (OFDMA) for downlink transmission to achieve high peak data rates in high spectrum bandwidth. LTE uses single carrier frequency division multiple access (SC-FDMA) for uplink transmission, a technology that provides advantages in power efficiency. LTE supports both FDD and TDD modes, with FDD, DL, and UL transmissions performed simultaneously in two different frequency bands, with TDD, DL, and UL transmissions performed at different time intervals within the same frequency band. LTE supports advanced adaptive MIMO, balance average/peak throughput, and coverage/cell-edge bit rate. Compared to 3G, significant reduction in delay over air interface can be supported in LTE, and it is suitable for real-time applications, for example, VoIP, PoC, gaming, and so on.

Spectrum is a finite resource and FDD and TDD system will support the future demand, which are shown in Figure 1.5. TDD spectrum can provide 100-150MHz of additional bandwidth per operator, TD-LTE spectrum with large bandwidth will be a key to operators future network strategy and one of the way to address capacity growth.

1.1.1 LTE Architecture

LTE is predominantly associated with the radio access network (RAN). The eNodeB (eNB) is the component within the LTE RAN network. LTE RAN provides the physical radio link between the user equipment (UE) and the evolved packet core network. The system architecture evolution (SAE) specifications defines a new core network, which is termed as evolved packet core (EPC) including all internet protocol (IP) networking architectures (Figure 1.6).

Evolved NodeB (eNB): Provides the LTE air interface to the UEs, the eNB terminates the user plane (PDCP/RLC/MAC/L1) and control plane (RRC) protocols. Among other things, it performs radio resource management and intra-LTE mobility for the evolved access system. At the S1 interface toward the EPC, the eNB terminates the control plane (S1AP) and the user plane (GTP-U).

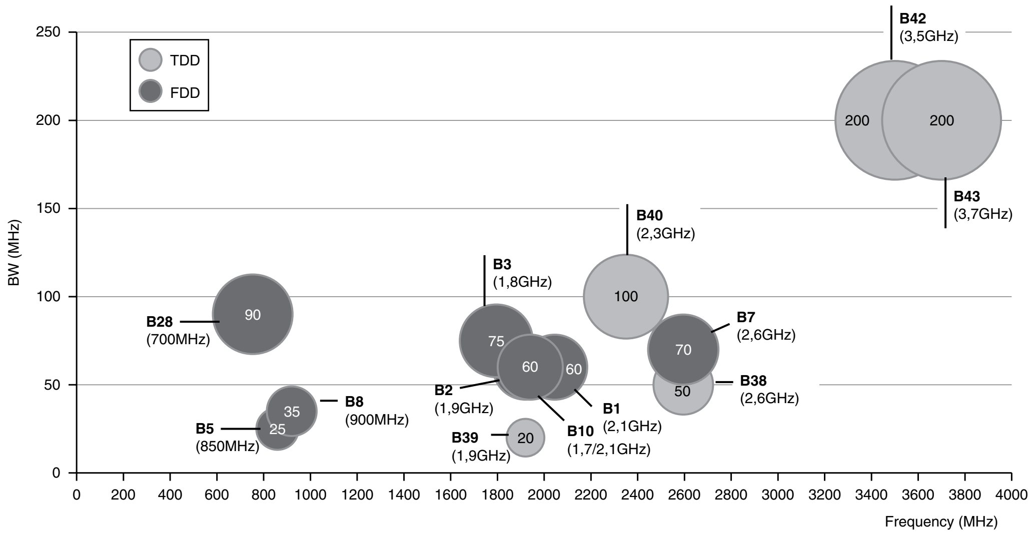


Figure 1.5 Spectrum of LTE.

Mobility Management Entity (MME): A control plane node responsible for idle mode UE tracking and paging procedures. The Non-Access Stratum (NAS) signaling terminates at the MME. Its main function is to manage mobility, UE identities, and security parameters. The MME is involved in the EPS bearer activation, modification, deactivation process, and is also responsible for choosing the SGW for a UE at the initial attach and at time of intra-LTE handover involving core network node relocation. PDN GW selection is also performed by the MME. It is responsible for authenticating the user by interacting with the home subscription server (HSS).

Serving Gateway (SGW): This node routes and forwards the IP packets, while also acting as the mobility anchor for the user plane flow during inter-eNB handovers and other 3GPP technologies (2G/3G systems using S4). For idle state UEs, the SGW terminates the DL data path and triggers paging when DL data arrives for the UE.

Packet Data Network Gateway (PDN GW): Provides connectivity to the UE to external packet data networks by being the point of exit and entry of traffic for the UEs. The PDN GW performs among other policy enforcement, packet filtering for each user and IP address allocation.

Policy and Charging Rules Function (PCRF): The PCRF supports policy control decisions and flow based charging control functionalities. Policy control is the process whereby the PCRF indicates to the PCEF (in PDN GW) how to control the EPS bearer. A policy in this context is the information that is going to be installed in the PCEF to allow the enforcement of the required services.

Home Subscription Server (HSS): The HSS is the master database that contains LTE user information and hosts the database of the LTE users.

1.1.2 LTE Network Interfaces

LTE network can be considered of two main components: RAN and EPC. RAN includes the LTE radio protocol stack (RRC, PDCP, RLC, MAC, PHY). These entities reside entirely within the UE and the eNB nodes. EPC includes core network interfaces, protocols, and entities. These entities and protocols reside within the SGW, PGW, and MME nodes, and partially within the eNB nodes.

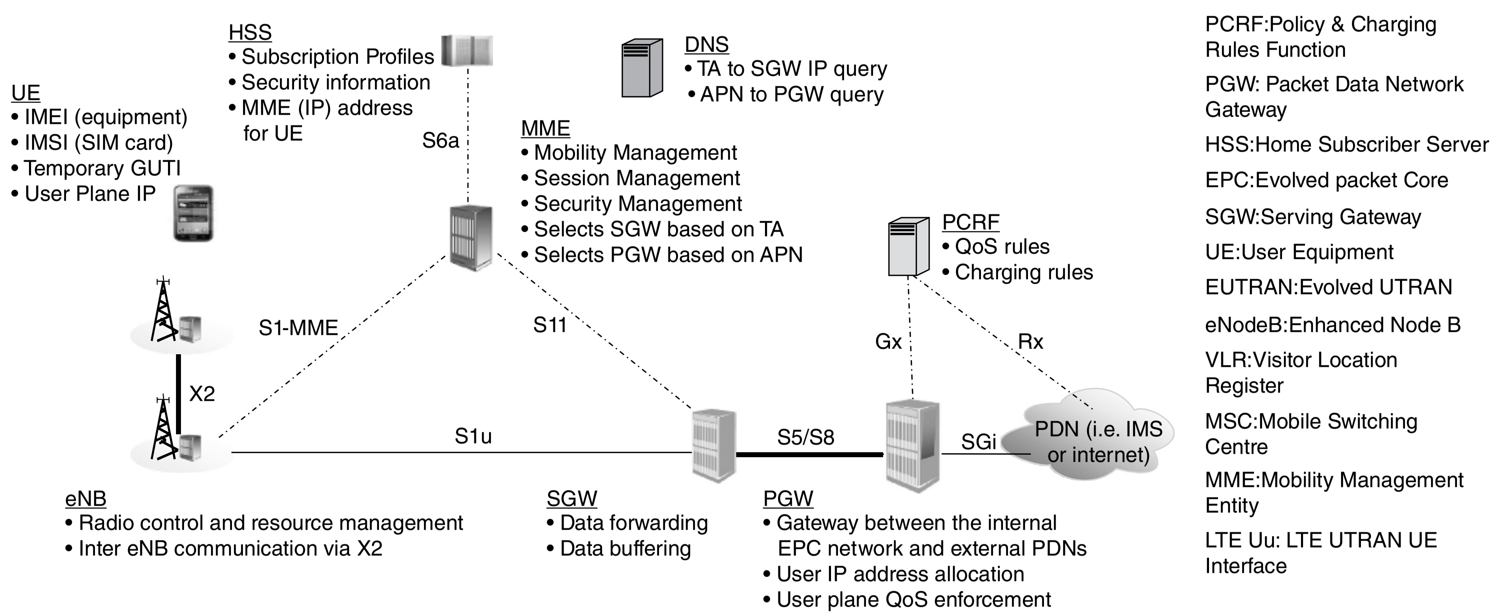


Figure 1.6 Nodes and functions in LTE.

Uu: Uu is the air interface connecting the eNB with the UEs. The protocols used for the control plane are RRC on top of PDCP, RLC, MAC, and L1. The protocols used for the user plane are PDCP, RLC, MAC, and L1. LTE air interface supports high data rates. LTE uses OFDMA for downlink transmission to achieve high peak data rates in high spectrum bandwidth. LTE uses SC-FDMA for uplink transmission, a technology that provides advantages in power efficiency.

S1: The interface S1 is used to connect the MME/S-GW and the eNB. The S1 is used for both the control plane and the user plane. The control plane part is referred to as S1-MME and the user plane S1-U. The protocol used on S1-MME is S1-AP on the radio network layer. The transport network layer is based on IP transport, comprising SCTP on top of IP. The protocol used on S1-U is based on IP transport with GTP-U and UDP on top.

The X2 interface is a new type of interface between the eNBs introduced by the LTE to perform the following functions: handover, load management, CoMP, and so on. X2-UP protocol tunnels end-user packets between the eNBs. The tunneling function supports are identification of packets with the tunnels and packet loss management. X2-UP uses GTP-U over UDP/IP as the transport layer protocol similar to S1-UP protocol. X2-CP has SCTP as the transport layer protocol is similar to the S1-CP protocol. The load management function allows exchange of overload and traffic load information between eNBs, which helps eNBs handle traffic load effectively. The handover function enables one eNB to hand over the UE to another eNB. A handover operation requires transfer of information necessary to maintain the services at the new eNB. It also requires establishment and release of tunnels between source and target eNB to allow data forwarding and informs the already prepared target eNB for handover cancellations.

NAS is a control plane protocol that terminates in both the UE and the MME. It is transparently carried over the Uu and S1 interface.

S6a: S6a interface enables transfer of subscription and authentication data between the MME and HSS for authenticating/authorizing user access to the EUTRAN. The S6a interface is involved in the following call flows, initial attach, tracking area update, service request, detach, HSS user profile management, and HSS-initiated QoS modification, and so on.

S11: Reference point between MME and SGW. This is a control plane interface for negotiating bearer plane resources with the SGW.

The above-mentioned LTE network interfaces are shown in Figure 1.7.

IP connection between a UE and a PDN is called PDN connection or EPS session. Each PDN connection is represented by an IP address of the UE and a PDN ID (APN). As shown in Figure 1.8, there are two different layers of IP networking. The first one is the end-to-end layer, which provides end-to-end connectivity to the users. This layers involves the UEs, the PGW, and the remote host, but does not involve the eNB. The second layer of IP networking is the EPC local area network, which involves all eNBs and the SGW/PGW node. The end-to-end IP communications is tunneled over the local EPC IP network using GTP/UDP/IP.

Moreover, in LTE, IDs are used to identify a different UE, mobile equipment, and network element to make the EPS data session and bearer establishment, which can refer to Annex "LTE identifiers" for reference; the summary of IDs is shown in Table 1.1.

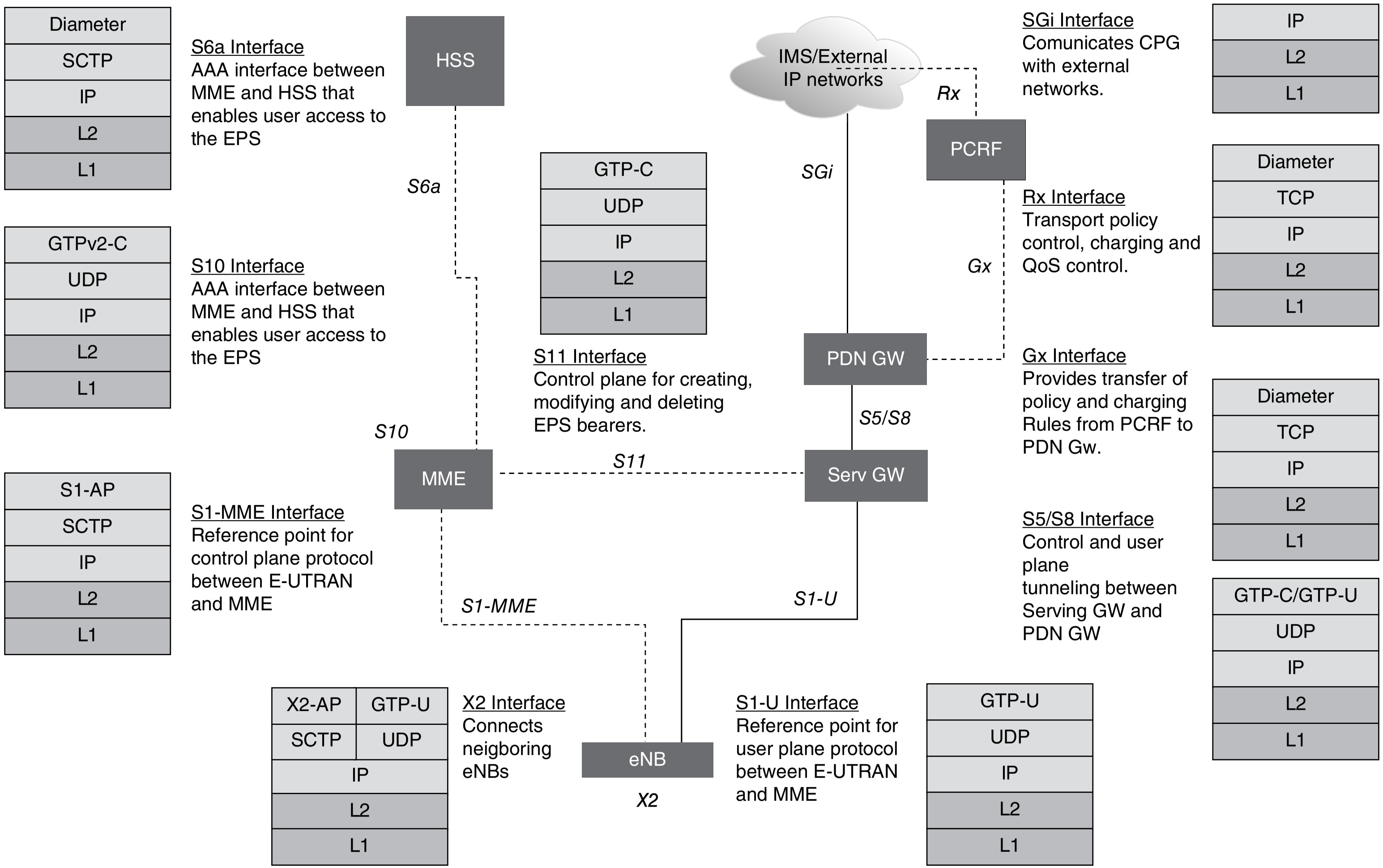


Figure 1.7 LTE network interfaces.

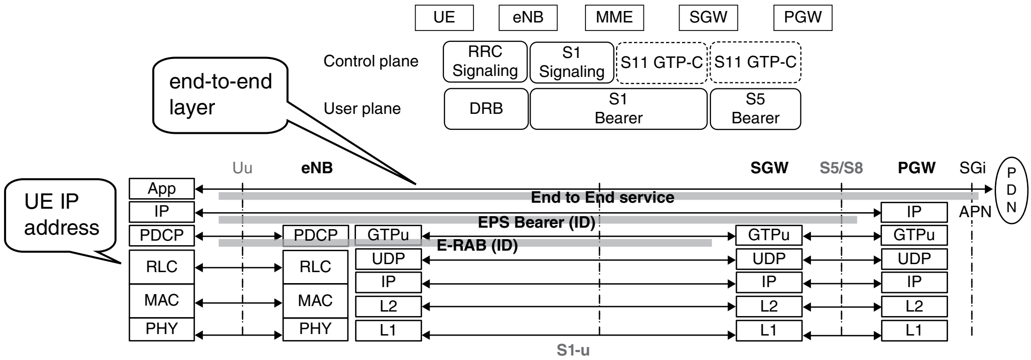


Figure 1.8 LTE-EPC control and data plane protocol stack.

Table 1.1 Classification of LTE identification.

Classification	LTE identification
UE ID	IMSI, GUTI, S-TMSI, IP address, C-RNTI, UE S1AP ID, UE X2AP ID
Mobile equipment ID	IMEI
Network element ID	GUMMEI, MMEI, Global eNB ID, eNB ID, ECGI, ECI, P-GW ID
Location ID	TAI, TAC
Session/bearer ID	PDN ID (APN), EPS bearer ID, E-RAB ID, DRB ID, LBI, TEID

1.2 LTE Services

LTE is an all packet-switched technology. The telephony service on LTE is a packet-switched mobile broadband service relying on specific support in LTE radio and EPC, which is needed to meet the expectations of telephony. On the other hand, the handling of voice traffic on LTE handsets is evolving as the mobile industry infrastructure evolves toward higher, eventually ubiquitous, and finally, LTE availability. Central to the enablement of LTE smartphones is to meet today's very high expectation for the mobile user experience and to evolve the entire communications experience by augmenting voice with richer media services. Voice solutions of LTE include VoLTE/SRVCC, RCS, OTT, CSFB, SVLTE, and so on. LTE radio and EPC architecture does not have a circuit-switched (CS) domain available to handle voice calls as being done in 2G/3G. The voice traffic in the LTE network is handled through different procedures. The first one, which is mainly used, still remains on the circuit switch network (e.g., 2G or 3G) by maintaining either parallel connection and registration on these network or by switching to them whenever a voice call is initiated or terminated. The second one, which is when the voice call stands over LTE, the voice service is named VoLTE or VoIMS when the IP multi-media system (IMS) service function is included.

Video in LTE is one of the most importanr services. The demand for video content continues to grow among data services. Web video traffic growth has accelerated, as the number of internet-enabled devices has increased and more people depend on the mobile internet.

Recently, a group of key operators, infrastructure, and device vendors announced a joint effort to facilitate the evolution of mobile communication toward RCS (rich communication suite). The core feature set of RCS includes the following services: enhanced phonebook, with service capabilities and presence enhanced contacts information; enhanced messaging, which enables a large variety of messaging options including chat and messaging history, and enriched call, which enables multimedia content sharing during a voice call. It is believed that RCS is a promising evolution in LTE, many operators have announced to support the RCS.

1.2.1 Circuit-Switched Fallback

The basic principle of circuit-switched fallback (CSFB) is that once originating or receiving a CS voice call by the UE connected over LTE, it will move to either GSM or UMTS network (fallback) where the call proceeds. One major requirement for the realization of CSFB is the overlay of LTE with GSM, UMTS, or both. It is the quickest implementation both at terminal and at network sides and is mandatory for international roaming scenarios. With CSFB, UE will attach to the network through LTE, MME will ask MSC to update UE location in its database, when the UE is operating in LTE (data connection) mode and when a call comes in, the LTE network pages the device. The device responds with a special service request message to the network, and the network signals the device to move to 2/3G to accept the incoming call. Similarly for outgoing calls, the same special service request is used to move the device to 2/3G to place the outgoing call.

CSFB for operator means very little investment since only few modifications are required in the network, additional interface (SGs) between MME and MSC is required shown in Figure 1.9. With basic CSFB implementation, the additional delay to set up the voice call is less than 1.3s to 3G or about 2.8s to 2G, which is acceptable from an end-user perspective. This delay is significantly reduced with the activation of PS handovers when falling back to 3G and of RRC release with 3GPP Rel 9 redirections to 2G/3G. The CSFB option offers complete services and feature transparency by enabling mobile service providers to leverage their existing GSM/UMTS network for the delivery of CS services, including prepaid and postpaid billing.

SGs interface is used to carry signaling to move the access network carrying the voice traffic from 2/3G to the LTE and from LTE to 2/3G. This interface maintains a connection between the MSC/VLR and the MME and its main role is to handle signaling and voice by SGsAP application.

Gn-C interface is the interface connecting the MME to the SGSN in the pre-Rel 8, it is replaced by the S3 for Rel 8 or later. This interface is required when a CSFB call is established to initial the signaling with SGSN. In case CSFB with PS handover the data established over the LTE will be carried over 2/3G network, the interface Gn-C or S3 is used to establish the signaling sessions with the SGSN to forward pending data over the LTE toward the 2/3G packet core. To forward the data from the PGW, an additional interface named Gn-U is required between the SGSN and the PGW in pre-Rel 8 and the S4 interface between the SGSN and the SGW in Rel 8 or later.

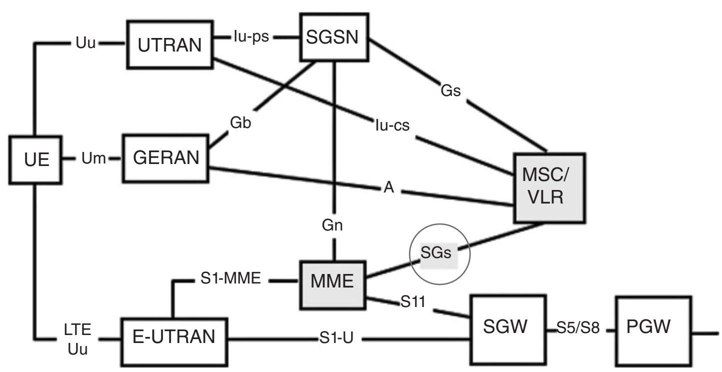


Figure 1.9 Standard architecture for CSFB.

Figure 1.10 Dual radio handsets.

CSFB is a single radio solution of handset, in order to make or receive calls, the UE must change its radio access technology from LTE to a 2G/3G technology, and uses network signaling to determine when to switch from the PS network to the CS network. The shortcoming is that someone on a voice call will not be able to use the LTE network for browsing or chatting, and so on. Except CSFB, dual-radio handsets (SVLTE) shown in Figure 1.10 support simultaneous voice and data— voice provided through legacy 2G or 3G network and data services provided by LTE. Dual-radio solutions use two always-on radios (and supporting chipsets), one for packet-switched LTE data and one for circuit-switched telephony, and as a data fallback where LTE is not available. The dual radio has the benefit in which simultaneous CS voice and LTE data is available; the drawback is the complexity from the device point of view, since more radio components are required increasing the cost, size, and power consumption. Dual-radio solutions also force the need for double subscriber registration leading to split legacy and LTE records in the subscriber data managers. As a matter of fact, lack of dual-radio eco-system for 3GPP markets and the top six main chipset vendors are addressing the 3GPP market with singe-radio terminal and CSFB, while the top chipset vendors for 3GPP2 markets are supporting dual-radio solution for the 3GPP2 market.

The above considerations have lead to a clear split in the market for early LTE support of voice services with mobile networks based on 3GPP technologies adopting CSFB, while 3GPP2 markets have adopted a dual-radio solution for early LTE deployments.

CSFB addresses the requirements of the first phase of the evolution of mobile voice services, which commercially launched in several regions around the world in 2011. CSFB has become the predominant global solution for voice and SMS inter-operability in early LTE handsets, primarily due to inherent cost, size, and battery life advantages of single-radio solutions on the device side. CSFB is the solution to the reality of mixed networks today and throughout the transition to ubiquitous all-LTE networks in the future phases of LTE voice evolution.

1.2.2 Voice over LTE

After CSFB, LTE voice evolution introduces native VoIP on LTE (VoLTE) along with enhanced IP multimedia services such as video telephony, HD voice and rich communication suite (RCS) additions like instant messaging, video share, and enhanced/shared phonebooks.

The voiceover LTE solution (VoLTE) is defined in the GSMA[1] *Permanent Reference Document* (PRD) IR.92,[2] based on the adopted one-voice profile (v 1.1.0) from the *One Voice Industry Initiative*. Video-related additions are described in GSMA IR.94.

1 At the 2010 GSMA mobile world congress, GSMA announced that they were supporting the one voice solution to provide voice over LTE. After that, industry aligned 3GPP based e2e solution for GSM equivalent voice services over LTE.
2 The VoLTE IR.92 is from October 2010 put in maintenance mode and only corrections of issues that may cause frequent and serious misoperation will be introduced.

VoLTE specifies the minimum requirements to be fulfilled by network operators and terminal vendors in order to provide a high-quality and interoperable voice over LTE service. The VoLTE solution is scalable and rapidly deployed, offering rich multimedia and voice services, seamless voice continuity across access networks, and the re-use of existing network investments including business and operational support assets. In terms of the operators, the deployment of VoLTE means that it is opened to the mobile wideband speech path of evolution. Also, VoLTE can offer a competitive advantage by providing a superior voice service quality with HD voice and video, shortening setup times for the calls and guaranteeing bit rate, and offering simultaneous LTE data together with the voice call. Finally, a richer end-user experience; to be able to provide end users the benefit of real-time communications can be another VoLTE attraction. Better multimedia, video-conferencing, or video chat while still maintaining a voice call, are all possible revenue opportunities of VoLTE. Introducing VoLTE on a standard-based IMS provides the service provider with a true converged network where services are available regardless of the access type network. Blending services with an IMS service architecture enables an operator to cost-effectively build integrated service bundles. VoLTE can evolve voice services into rich multimedia offerings, including HD voice, video calling, and other multimedia services (i.e., start a voice session, add and drop media such as video, and add callers, presence) available anywhere on any device, combining mobility with service continuity.

VoLTE is an advancement from today's voice and video telephony to full-fledged multimedia communication to utilize the full potential of LTE and to improve customer experience. The IP-based call is always anchored in IMS core network to carry and establish a voice call over an LTE network. Now, in both 3GPP and 3GPP2 markets, there is a clear consensus to adopt the IMS-based VoLTE solution for the LTE deployments.

Two transport modes are also used on the network and determines the quality of the voice call over an IP network. The VoIP's best effort, mainly over the internet and based on some widely deployed applications, such as Skype, Google talk, and MSN, uses this mode with no guarantee of the quality. Other technology such as LTE propose to carry the VoIP with the guarantee of the quality of this call over the end-to-end network. For VoLTE, the installed solution aims at being partially compliant with GSMA PRD IR.92.[3] One voice was an effort to use already-defined standards to specify a mandatory set of functionality for devices, the LTE access network, the evolved packet core network, and the IP multimedia subsystem in order to define a voice and SMS over LTE solution using an IMS architecture. Some VoLTE handsets are already commercial including the features such as emergency call, location based services, and so on.

In case VoLTE through IMS is the mode used, two connections are required with the LTE network—Rx interface between the P-CSCF and the PCRF and the Gx interface between the PCRF and the PGW for dynamic PCC rules. The Gm interface is a virtual interface established between the SIP application on the end user and the P-CSCF function of the IMS network where it is connected (Figure 1.11).

Along with VoLTE introduction, 3GPP also standardized Single Radio Voice Call Continuity (SRVCC) in Rel 8 specifications to provide seamless continuity when an UE handovers from LTE coverage (E-UTRAN) to UMTS/GSM coverage (UTRAN/GERAN). With SRVCC, which is depicted in Figure 1.12, the calls are anchored in IMS network while UE is capable of transmitting/receiving on only one of those access networks at a given time. SRVCC protocol evolution have different types according to the function. There are bSRVCC (before alerting

3 Complementary scenarios are also beign defined in the VoLTE profile extension (IR.93) to cope with the cases where LTE coverage needs to be complemented with existing WCDMA/GSM CS coverage.

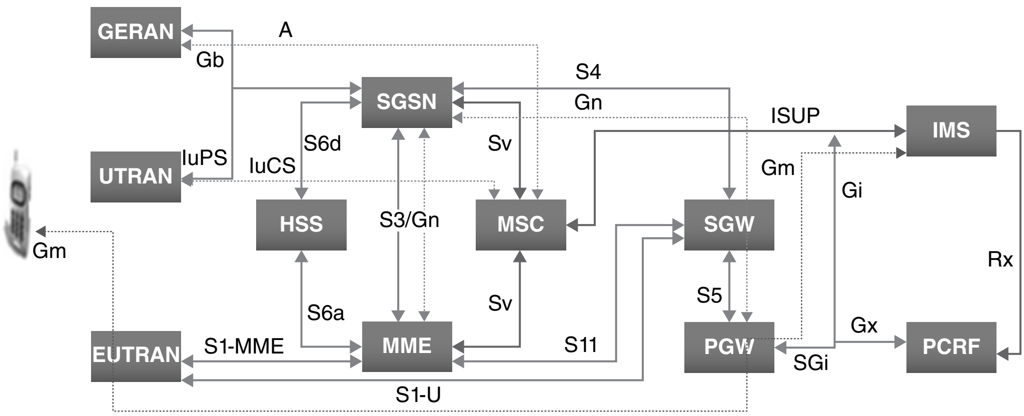


Figure 1.11 Standard architecture for VoLTE.

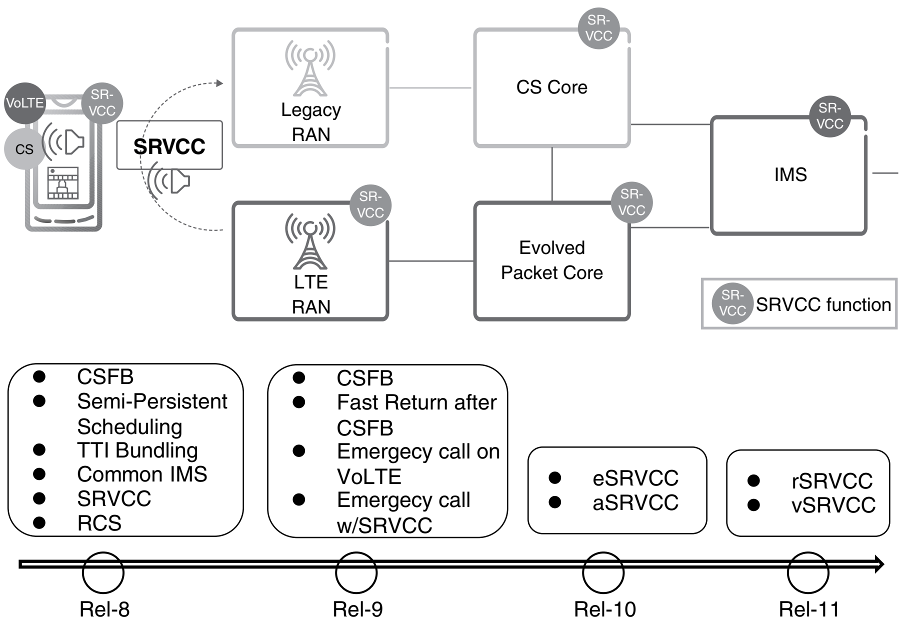


Figure 1.12 SRVCC and evolution.

SRVCC), aSRVCC (alerting phase SRVCC), vSRVCC (video SRVCC), and vSRVCC (reverse SRVCC, HO 3G/2G → LTE).

Up to now, VoLTE launches are taking place in Korea, the United States (AT&T, T-Mobile, Verizon), Russia (MTS), and Asia (NTT Docomo, SingTel, M1, Starhub, HKT). T-Mobile U.S. launched VoLTE in Seattle on May 22, 2014. AT&T launched in three markets on May 23, 2014 with "crystal clear conversations." SingTel launched on May 31, 2014 in Singapore using 4G Clear Voice. In 2015 and 2016, more and more countries launched VoLTE, like China, Canada, France, and Denmark.

1.2.3 IMS Centralized Services

In a CS network telephony, services are provided by the MSC (based on the subscription data in the HLR).

In IMS telephony, services are provided by the telephony application server. Multiple service engines introduce synchronization problems and differences in user experience. IMS centralized services avoids these problems by assuming that only one service engine will be used.

IMS plays an essential role in IMS centralized services. The UE performs SIP (session initiation protocol) registration with the IMS network. IMS-AKA (IMS-authentication and key agreement) procedures are followed for authentication. Integrity protection, whereby integrity of SIP signaling messages is ensured, is mandatory. The use of ISIM (IP multimedia services identity module) or USIM (UMTS subscriber identity module) is required during the IMS authentication. SIP signaling messages are ASCII text messages and could thus be quite large. Hence, signaling compression is mandatory to reduce the bandwidth requirements, especially for over-the-air transmission.

IMS centralized services (ICS) enable the use of the IMS telephony service engine for originating and terminating services regardless if a UE is connected via a LTE PS access network or connected via a GSM/WCDMA CS access network. For terminating calls, ICS determines the access network currently in use by a UE to deliver the call via the correct access network. ICS requires an IMS service centralization and continuity application server.

1.2.4 Over the Top Solutions

At the same time, there are already a number of applications providing over the top (OTT) voice service on smartphones, which can be used over Wi-Fi connection but also over cellular networks. OTT application is completely transparent to network and also out of operators' control. OTT services are those provided without special consideration at the network level (i.e., no special treatment with respect to QoS). Examples of these types of services are YouTube, Vimeo, and DailyMotion, which are very popular today. Skype and GoogleTalk have nearly a billion registered users worldwide. Apple has sold countless iPhones and iPads, many of which are capable of FaceTime video calling. These services are provided directly by content providers (and usually over content delivery networks), generally without any arrangement with the network providers sitting between the content and its consumers. Nowadays, some OTT solutions, such as Skype and FaceTime, often come preinstalled on smartphones, and as these devices become much more widespread, the adoption of OTT solutions for video-calling services will also increase. LTE supports high bandwidth, low latency, always online, all IP and other characteristics, it is convenient for the development of OTT. OTT application providers have delivered very popular voice, video, messaging, and location services that are shifting consumers' attention and usage. In addition, while OTT players currently generate revenue using the operator's network for service delivery, the operator itself doesn't gain any associated increase in revenues. The Figure 1.13 shows MoS performance based on data from the South Korean market's most OTT-friendly operator.

In the future, the proportion of OTT voice may be more and more high, especially in the area of long distance calls, as these solutions are familiar to subscribers and have driven user expectations. However, a fully satisfactory user experience cannot be provided by OTT solutions, as there are no QoS measures in place, no handover mechanism to the circuit-switched network, no widespread interoperability of services between different OTT services and devices, and no guaranteed emergency support or security measures. Consequently, the adoption of OTT clients is directly dependent on mobile broadband coverage and the willingness of subscribers to use a service that lacks quality, security, and flexibility. For example, with VoLTE, using

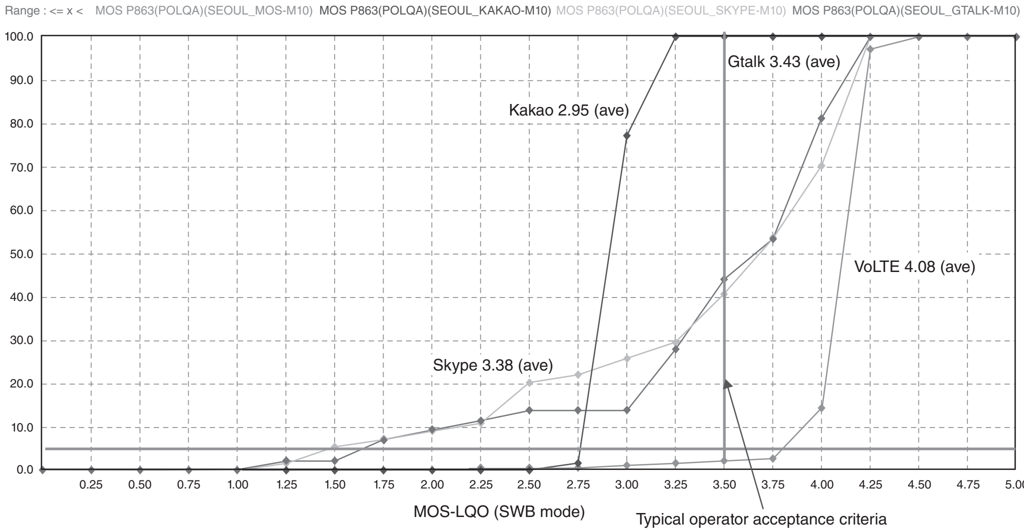


Figure 1.13 MoS-LQO (SWB, super wideband AMR mode).

SRVCC, the voice communication can be maintained when the user is moving out of LTE coverage versus OTT call might drop in this case.

Voice over LTE delivers very high spectral efficiency, OTT solutions that do not benefit from radio performance enhancements implemented specifically for VoLTE, for example, dynamic scheduling, RoHC, TTI bundling, packet segmentation, and admission control. VoLTE also provides better quality than OTT services and narrowband CS calls, as well as improved call setup time, HD voice quality, E2E QoS guaranteed, longer battery life, emergency call availability, and more efficiency than OTT in terms of network resource consumption. Also, rich multimedia voice could be offered directly enabled on top of VoLTE as well, delivering high user experiences beyond voice and SMS, providing to consumers instant messaging, chat, video, and file sharing.

1.2.5 SMS Alternatives over LTE

Basically two options are available for SMS, SMS without IMS, and SMS with IMS. When the device registers as an IMS user, then an incoming SMS will be directed to IMS and delivered via IP.

For SMS using CSFB (SMS over SGs), UE can send and retrieve SMS using NAS signaling over LTE. This solution requires SGs interface between CS core and EPC to transport SMS to/from UE. The SGs interface must support mobility management and paging procedures between the EPS and the CS domain. All the procedures are based on the SGs procedures.

SMS over IP is a solution to transport SMS over any PS access using IMS and is mandated to be supported by VoLTE terminals (Figure 1.14).

If not registered in IMS, and CS attached, then incoming SMS will be delivered to CS domain. Both options support UE-originated and UE-terminated SMSs. In the non-IMS approach, the MME needs to support the SGs interface toward MSC, MSC pages the UE over LTE via this new channel for terminating SMS. The UE includes the SMS inside a NAS signaling message, and the MME forwards the SMS to the MSC. SMS is supported in both idle mode and connected mode. Of course, if the UE is in idle mode, it is needed to first establish connectivity between the UE and the E-UTRAN and between the UE and the MME. The UE needs to send

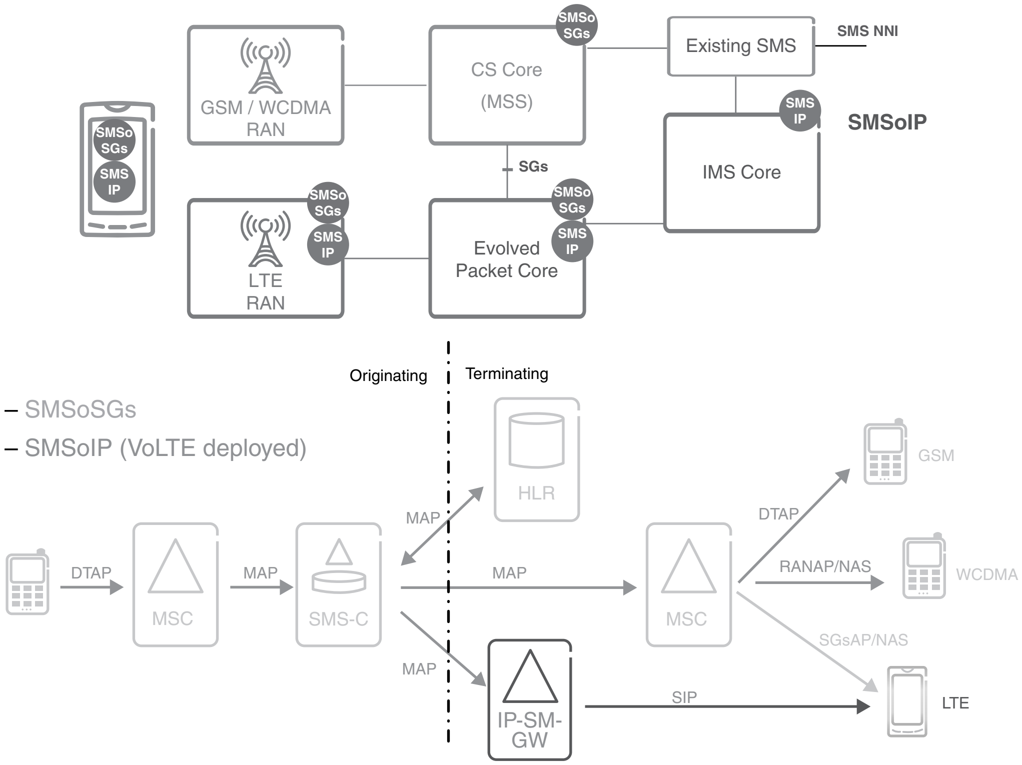


Figure 1.14 SMSoSGs and SMSoIP.

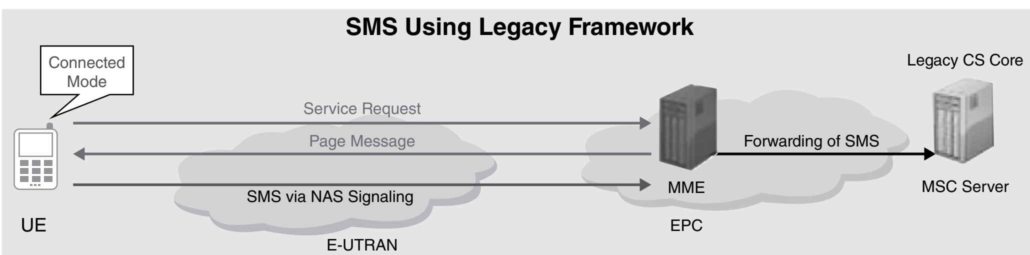


Figure 1.15 SMS using legacy framework.

a service request message to the MME to exit the idle mode. Once the MSC receives the service request message from MME, it will send the SMS via SGs interface to the MME, which will tunnel the short message to the UE.

For a UE-terminated SMS, a page message would be sent to the UE to get the UE out of the mode. If the UE is in connected mode, it already has all the links established. This will remove the extra service request/paging type signaling exchanges. The UE and the MME can directly place an SMS in a NAS signaling message (Figure 1.15).

In the case of SMS using IMS, the UE needs to implement the functions of an SMS- over-IP sender and an SMS-over-IP receiver. The IMS core network performs functions of an IP short message gateway (IP-SM-GW). For a receiver to get the SMS, the receiver needs to do IMS registration and indicate its capability to receive traditional short messages (Figure 1.16).

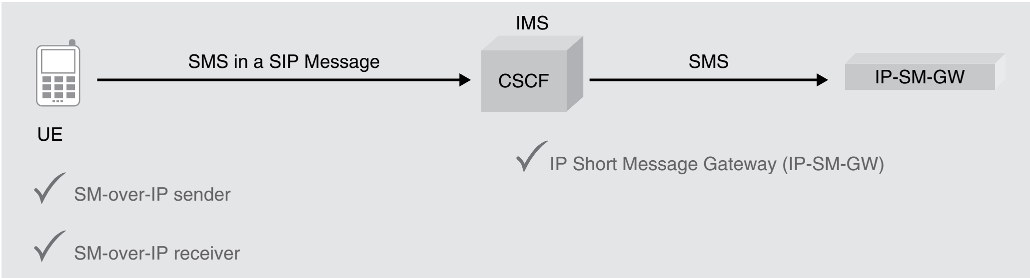


Figure 1.16 SMS using IMS.

1.2.6 Converged Communication

Converged communication is network convergence, media convergence, IT and CT convergence, as well as the convergence of communication and social networks. It will leverage cloud computing, mass data, and other emerging technologies to help drive the next paradigm shifts of the RCS initiative.

RCS IP call is derived from VoLTE specifications and as such can be offered over any IP connectivity as an extension to VoLTE. Its primary use case is when VoLTE is not available or not wanted by the user for voice and/or video calls, for example, when roaming or while only CS service is available for voice. RCS IP call uses the same control and media plane as VoLTE but adds RCS-specific policies. Unlike VoLTE, a RCS IP call does not rely on specific support from the access regarding QoS and mobility; hence operators can enable it over Wi-Fi, LTE, and even 3G without having VoLTE-specific support in those accesses.

Historically, RCS and VoLTE have been standardized separately from each other, meaning that coexistence aspects have not always been considered. However, that has changed since 2011. RCS, and in particular, RCS 5.1, takes coexistence between VoLTE and RCS into consideration. VoLTE and RCS complement each other, where VoLTE is the basis for primary real-time communication, and RCS provides enrichment and creates added value in the overall communication experience. RCS can also be deployed with CS voice before VoLTE deployment. So, RCS and VoLTE make use of one number for any service. RCS extends reachability outside the LTE access technology and opens up communication for both multi access and multi device usage.

Many operators have different plans for introduction of VoLTE and RCS in their networks. In Europe we see RCS first, while in North America and Korea, VoLTE has already been launched. Depending on the strategy, different evolution paths for coexistence will exist, RCS first, VoLTE first, or a combination from day one (Figure 1.17).

1.3 LTE Key Technology Overview

For LTE air interface technology, the choice of multiple-access schemes was made in December 2005, with orthogonal frequency division multiple access (OFDMA) being selected for the downlink, and single carrier-frequency division multiple access (SC-FDMA) for the uplink. In LTE downlink, OFDM will be used to schedule the UEs differently in time and frequency. With these techniques the UEs can be scheduled and receive user data in the downlink. In the uplink, SC-FDMA is similar to downlink and it is possible to schedule the UEs differently in time and frequency. LTE supports FDD and TDD mode. FDD and TDD LTE mode can be combined (depends on UE capabilities) in the same physical layer.

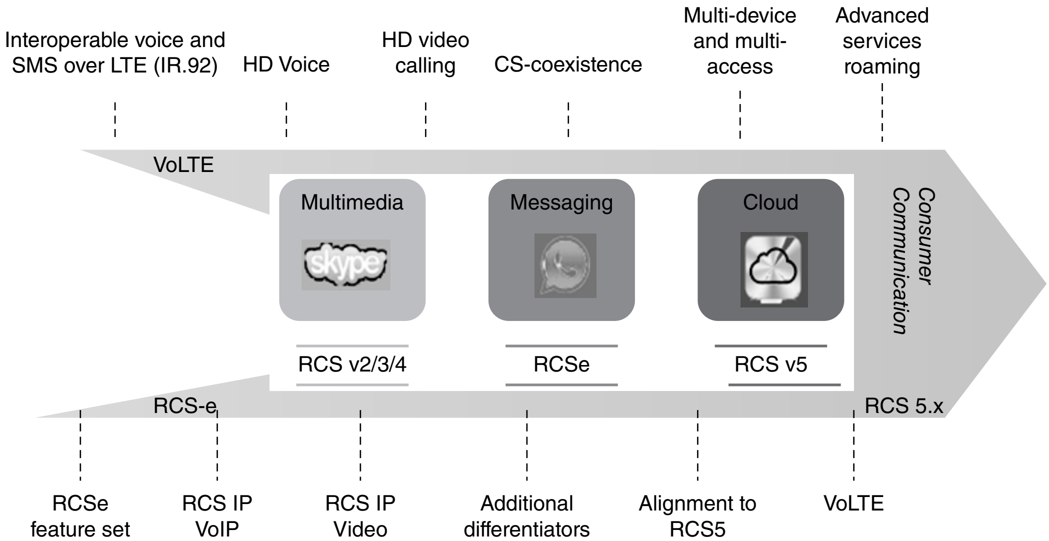


Figure 1.17 RCS is continuously innovating and service model are always changing.

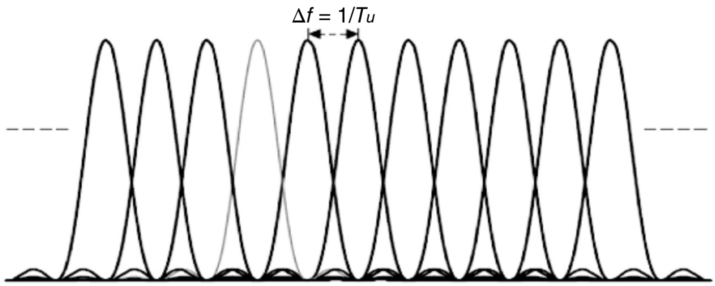
1.3.1 Orthogonal Frequency Division Multiplexing

OFDMA is the multi-access technology related with OFDM, the combination of TDMA and FDMA essentially. OFDMA has high spectrum utilization efficiency due to orthogonal subcarriers need no protect bandwidth, support frequency link auto adaptation and scheduling, and easy to combine with MIMO, but it needs strict requirement of time-frequency domain synchronization and high PAPR.

At the transmitter the coded and modulated data stream is split up to a number of sub-streams. The number of sub-streams can range from typically 12 (one resource block) and up to 1200 (100 resource blocks at 20MHz bandwidth). Each stream is fed into the IFFT block and transformed into a corresponding subcarrier. Each subcarrier in OFDM is 15 kHz. One subcarrier carries one OFDM symbol. With this subcarrier size, symbol time is 66.7μs, which should be much longer than the delay spread in order to keep the ISI (inter-symbol interference) low. This choice is based on the average radio channel delay spread (a measure of the radio channel time dispersion) and the coherence time (a measure of how slow the radio channel changes), which should fulfill "Delay spread << Symbol time < Coherence time." Also, the cyclic prefix should be longer than the expected delay spread in order to completely remove ISI. However, if the symbol time is too long (i.e., longer than the coherence time), the radio channel will change considerably during one symbol (Figure 1.18). This would lead to inter carrier interference (ICI).

The biggest drawback of OFDMA is that it suffers from high peak-to-average power ratio (PAPR). The higher the PAPR, the higher output power back-off (OBO) is needed to avoid clipping of transmitted signal, which in turn results in higher adjacent carrier leakage ratio (ACLR) and increased error vector magnitude (EVM). High OBO decreases power efficiency of the transmitter.

Similar to OFDMA, the SC-FDMA physical resource can be seen as a time-frequency grid with the additional constraint that a resource assigned to a UE must always consist of a set of consecutive subcarriers in order to keep the single-carrier property of the uplink transmission that can release the UE PA limitation caused by high PAPR. The SC-FDMA subcarrier spacing equals Δf=15 kHz and resource blocks, consisting of 12 subcarriers in the frequency domain, are defined also for the uplink.



OFDM subcarrier spacing.

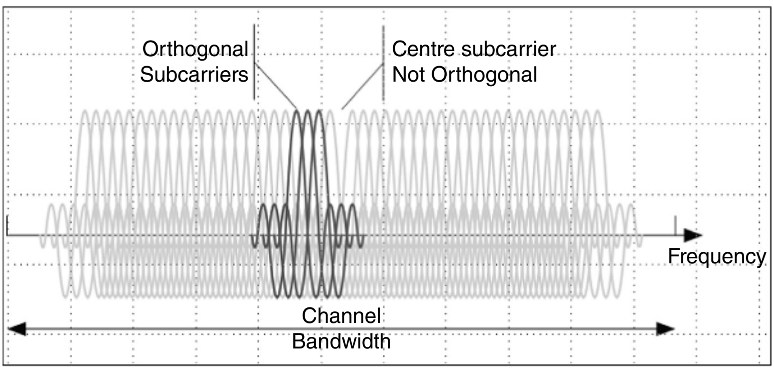


Figure 1.18 OFDMA.

1.3.2 MIMO

MIMO (multiple input multiple output) technology is a standard feature of LTE networks, and it is a major piece of LTE's promise to significantly boost data rates and overall system capacity. MIMO systems use more than one transmit antenna (Tx) to send a signal on the same frequency to more than one receive antenna (Rx). MIMO works under rich scattering conditions; signals from different TX take multiple paths to reach the UE at different times. In order to achieve promised throughputs in LTE systems, operators must optimize their networks' multipath conditions for MIMO, targeting both rich scattering conditions and high SNR for each multipath signal.

MIMO builds on Single Input Multiple Output (SIMO), also called receive diversity that have been around for decades, as well as Multiple Input Single Output (MISO), also called transmit diversity, used in most advanced cellular networks today. SIMO and MISO are both techniques seeking to boost signal to noise ratio (SNR) in order to compensate for signal degradation. MIMO can work as a combination of SIMO and MISO techniques, resulting in even greater SNR gains, further boosting coverage and data rates. It is the well-known techniques named transmit diversity and spatial multiplexing.

The other technique in LTE MIMO is called beamforming. It is only within the last decade that there has been significant interest in such antenna arrays for cellular systems. Beamforming generally requires a different antenna configuration from MIMO operations, which all the transmission paths are closely correlated, and each signal is affected by the transmission medium in a similar way. In this case, the multiple antennas can operate as though they were a single, high-power antenna with its main lobe illuminating a specific area. Beamforming aims to concentrate the available bandwidth in the targeted area, improving the quality of transmission or the signal to interference plus noise ratio (SINR). TD-LTE can make particularly good use of beam-forming antennas, as the uplink and downlink are duplexed in the time domain and transmitted and received signals at the same frequency.

1.3.3 Radio Resource Management

Radio resource management (RRM) includes assignment, re-assignment, and release of radio resources. Figure 1.19 shows time scale for different RRM mechanism in LTE network.

Radio bearer control is responsible for the establishment, maintenance, and release of radio resources associated with specific radio bearers. The radio bearer control function must consider the overall resources situation and the quality of service (QoS) requirements of in progress sessions when admitting new sessions. The radio bearer control function must also maintain the quality of existing sessions when conditions change due to environmental and mobility activity.

Admission and congestion control is a cell-based operation applied to both uplink and downlink. It is one of the key RRM functions. It is responsible for maximizing the radio resource utilization while meeting QoS requirements of new and existing sessions by intelligent admission or rejection of new radio bearer requests or modification of existing bearers. The task of admission control is to admit or to reject the requests for establishment of radio bearers (RB). Admission control is performed when there are new incoming calls or incoming handover attempts. The system resource limitations and QoS satisfaction ratio are the main considerations for admission control. The allocation/retention priority (ARP) can be used to classify gold, silver, and bronze categories with different admission control thresholds. ARP is an attribute of services and is inherited from evolved packet core.

Congestion control is critical to maintain the system stability and deliver acceptable QoS when the system is in congestion. The load measures include the power, the available physical resource block at the air interface, and the transmission resource usage at S1-U interface. Congestion control method including release low-priority services to alleviate the overloaded system, or GBR downsizing by sacrificing the quality of GBR services slightly but still maintaining acceptable quality.

Connection mobility control is responsible for the management of radio resources during active or idle mode mobility of the UEs. It handles the communication of mobility related parameters to the UE, and makes handover decisions based on both radio conditions and network-loading conditions.

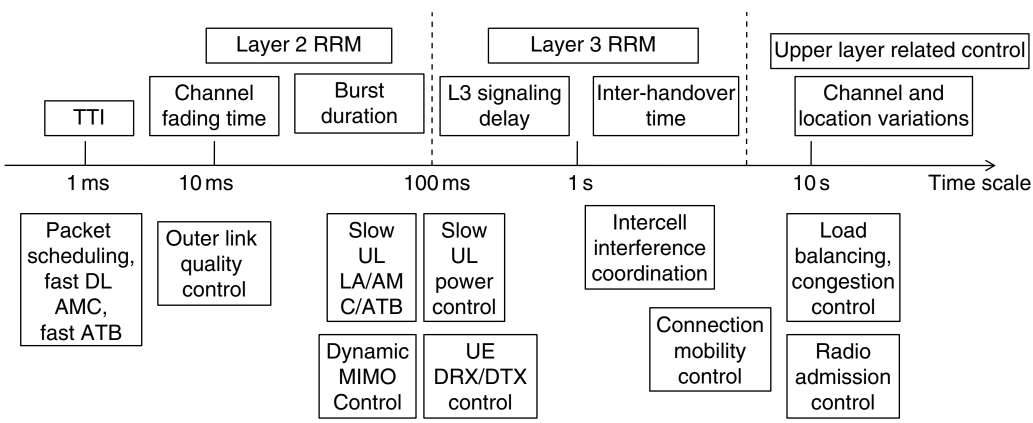


Figure 1.19 Time scale for different RRM mechanism.

In some situation of commercial LTE network, some serving cells have high load but other neighbor cells' loads are low because of the differentiation of UE services. Under this condition, it can trigger load balancing algorithm. Load balancing can also be triggered during network attach as part of initial network entry or idle exit. However, eNB will only load balance to different carrier on same sector for network attach, idle exit process.

In addition, inter-RAT radio resource management is responsible for resource management related to handovers with or redirection to a cell from a different radio technology. Handovers may be performed due to resource issues, radio conditions, or network preference specified by the operator.