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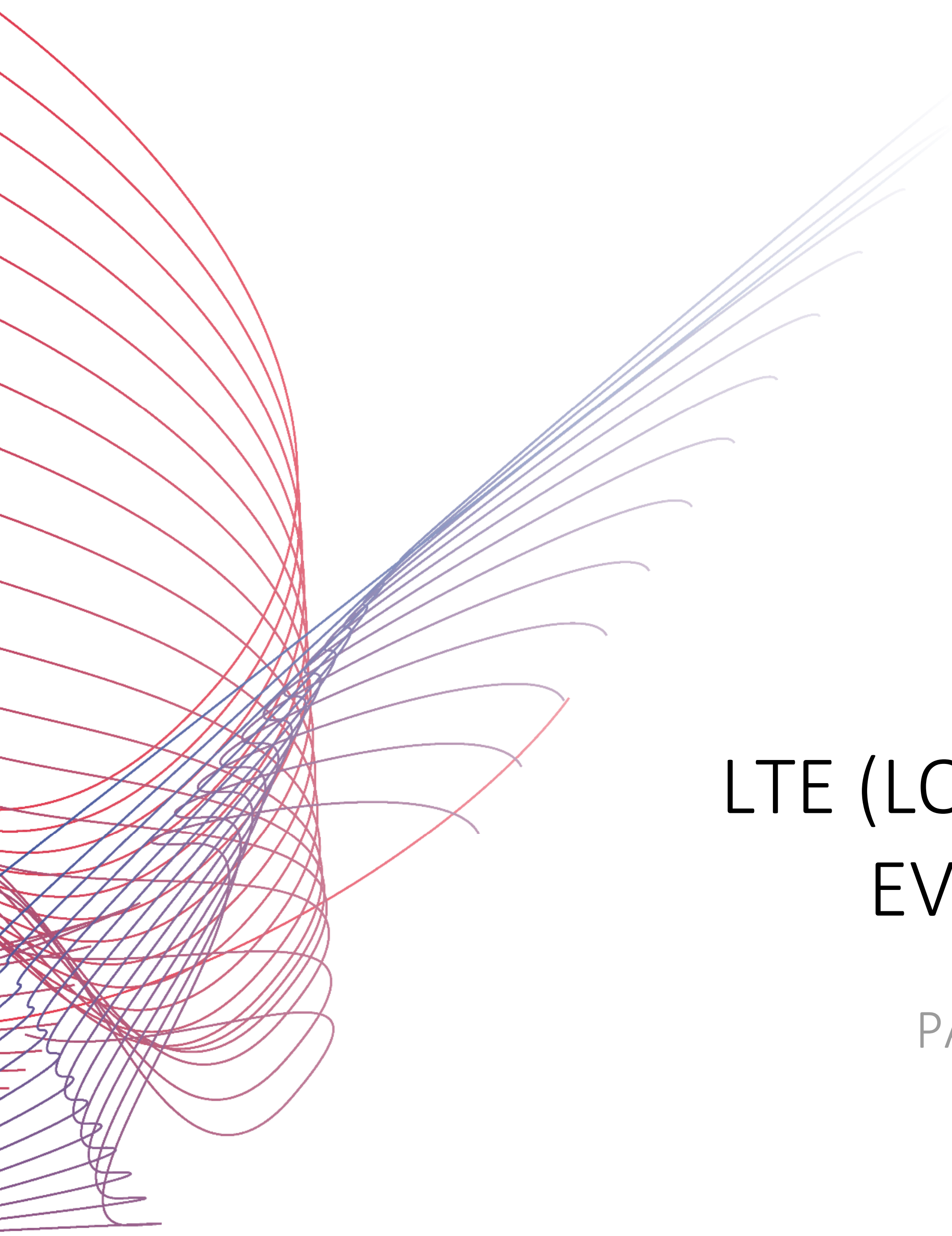

LTE (LONG TERM EVOLUTION)

PAVEL BEZPALEC

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Explanatory Notes

$E = m \cdot c^2$	Definition
	Interesting
i	Note
	Example
	Summary
+	Advantage
–	Disadvantage

ANNOTATION

Long term evolution (LTE) is the next step forward in cellular 3G services. LTE technology is a based on a 3GPP standard that provides for a downlink speed of up to 150 megabits per second (Mbps) and an uplink speed of up to 50 Mbps. Fixed wireless and wired standards are already approaching or achieving 100 Mbps or faster, and LTE is a way for cellular communications to operate at that high data rate.

OBJECTIVES

Overall knowledge about LTE technology.

LITERATURE

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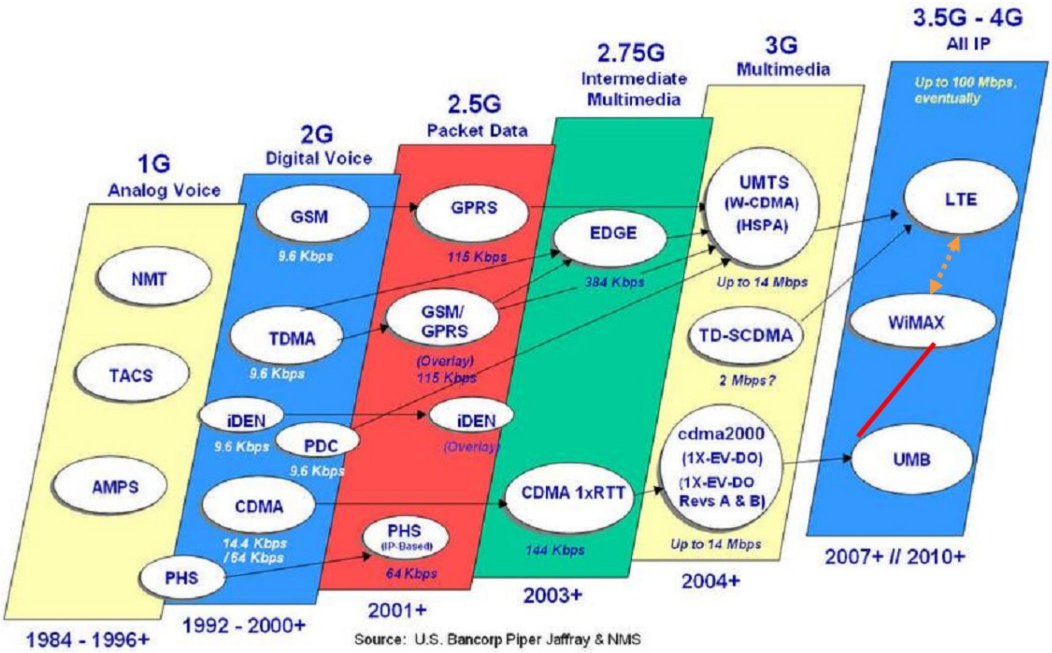
Index

1 Overview of Mobile Technologies for Voice and Data Transport

LTE stands for *Long Term Evolution* and is a registered trademark owned by **ETSI** (*European Telecommunications Standards Institute*) for the wireless data communications technology and a development of the GSM/UMTS standards. However other nations and companies do play an active role in the LTE project. The goal of LTE was to increase the capacity and speed of wireless data networks using new DSP (digital signal processing) techniques and modulations that were developed around the turn of the millennium. A further goal was the redesign and simplification of the network architecture to an IP-based system with significantly reduced transfer latency compared to the 3G architecture. The LTE wireless interface is incompatible with 2G and 3G networks, so that it must be operated on a separate radio spectrum.

In order to understand the major trends of evolved 3G architecture, it is necessary to look at the main steps of wireless network evolution, starting with 2G networks.



Wireless technology migration

1.1 2G Architecture

2G GSM cellular networks were initially designed for voice and circuit-switched services. For that reason, the architecture of such networks was comparatively simple and comprises two main parts:

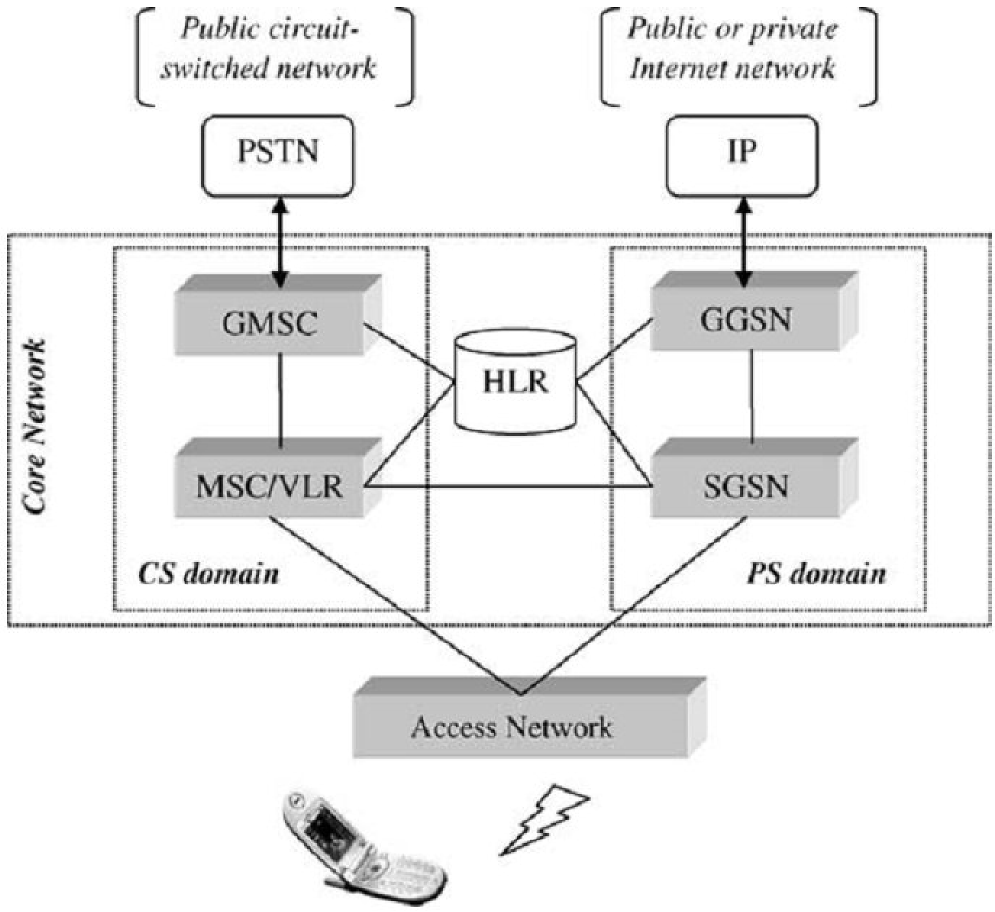
- The Access Network part, which includes the radio interface as well as the network nodes and interfaces supporting radio-related functions. In initial 2G GSM systems, the radiointerface was specifically designed and optimized for voice or low bit rate circuit data transmission.

- The CS part – or circuit-switched core network domain providing circuit services support (this includes call setup, authentication and billing) and inter-working with classical **PSTN** (*Public Switched Telephone Network*).

With the emergence of IP and Web services, 2G GSM networks eventually evolved to efficiently support packet data transmission:

- The Access Network part was partly redesigned to support packet transmission and shared resource allocation schemes – as for GPRS and EDGE evolutions.

- A new Core Network domain (PS for Packet Switched) was added, in parallel to the CS domain. This new domain has the same role as the CS domain, meaning support for packet transmission (including authentication and billing) as well as inter-working with public or private Internet (or IP) networks.



2G Dual Core Network Model

The CS domain is composed of a **MSC/VLR** (*Mobile Switching Center/Visitor Location Register*) responsible for end-to-end call setup and in charge of maintaining user location information (such information is typically used to page a user terminal in order to establish user-terminated communication sessions). The **GMSC** (*Gateway MSC*) is a specific type of MSC, as being the gateway switch responsible for PSTN inter-working.

The PS domain is composed of the **SGSN** (*Serving GPRS Support Node*), which basically plays the role of a MSC/VLR for the packet domain, and the **GGSN** (*Gateway GPRS Support Node*), which is equivalent to the GMSC for inter-working with external packet networks.

PS and CS domains may possibly be linked together in order to maintain consistent user location information between the two domains and therefore reduce the amount of radio and network signalling.

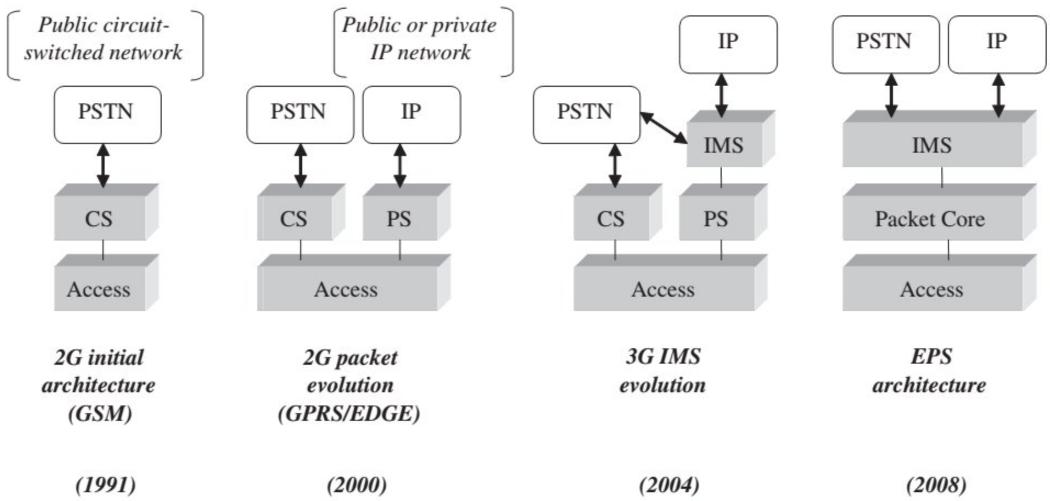
In addition to the domain-specific nodes, the Core Network also contains the **HLR** (*Home Location Register*), accessed by both the CS and PS domains. The HLR is a key part of the network architecture, containing all information related to user subscription.

1.2 3G IMS Evolution

From a system overview, initial 3G UMTS network architecture was more or less the same as the 2G, as it included both circuit and packet Core Networks. Eventually, a new domain was added on top of the PS domain: the **IMS** (*IP Multimedia Subsystem*).

The main objective of IMS was to allow the creation of standard and interoperable IP services (like *Push-To-Talk*, *Presence* or *Instant Messaging*) in a consistent way across 3GPP wireless networks. The interoperability of IMS-based services comes from the fact that IMS is based on flexible protocols like **SIP** (*Session Initiation Protocol*) developed by the IETF.

In addition, the IMS standard offers **VoIP** (*Voice over IP*) support and provides interworking with classical PSTN through signalling and media gateways.



Evolution of newotk architecture

As presented in Figure, the CS domain was still part of the 3G Core Network architecture, along with the PS/IMS structure. The main reason for keeping the CS domain was motivated by the need to support the – still dominant – circuit-switched voice services and H324M-based video-telephony support.

Although IMS was presented as an interesting step towards service integration, legacy network operators refrained from widely deploying and using it as a common platform for all services (including voice, real-time and nonreal-time services) because of the lack of support for voice services‘ seamless mobility between existing CS-based networks and IMS.

1.3 EPS Architecture

Evolved UMTS networks have a clear objective to integrate all applications over a simplified and common architecture. The main components of EPS architecture are thefollowing:

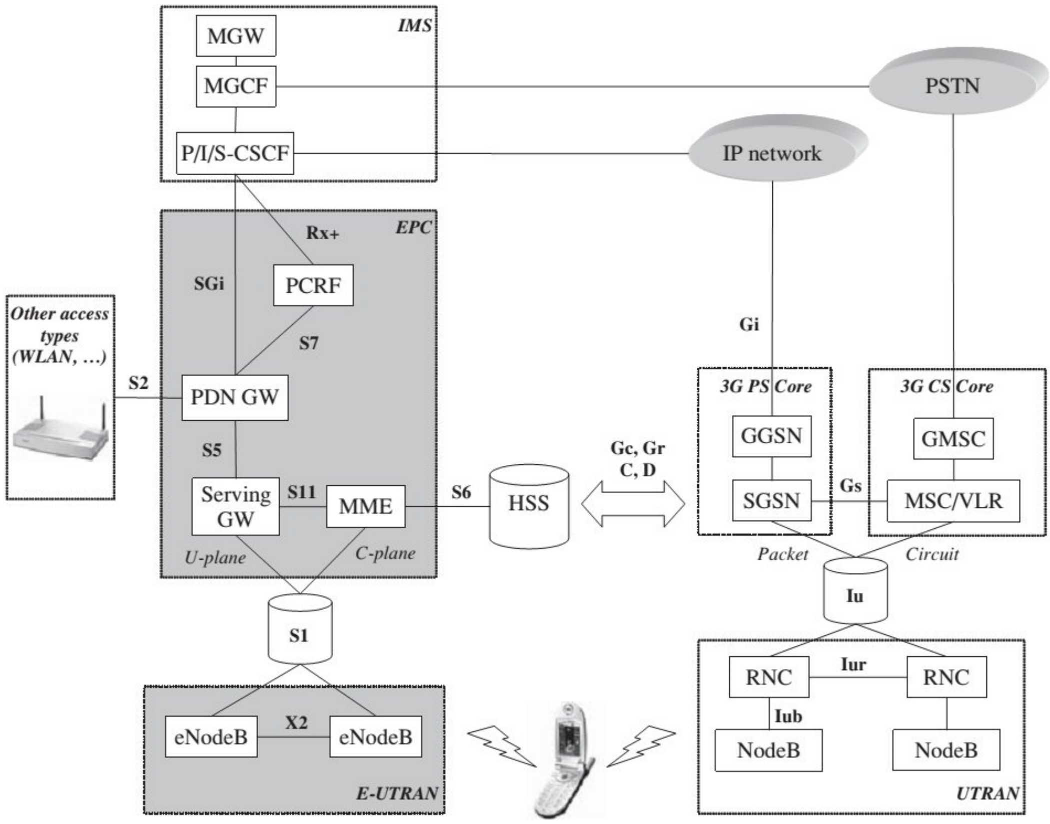
- A packet-optimized Access Network which can efficiently support IP-based nonreal-time services as well as circuit-like services requiring constant delay and constant bit rate transmission.
- A simplified Core Network, composed of only one packet domain, supporting all PS services (possibly IMS-based) and inter-working capabilities towards traditional PSTN.

The CS domain is no longer present, as all applications (including the most real-timeconstraining ones) are supported over the PS domain. This obviously requires specific gateway nodes – part of the IMS architecture – so that IP traffic is converted to PSTN circuit-switched-based transport.

As a consequence of this network simplification, specific efforts have been produced in the scope of Evolved UMTS standardization activity in order to maintain voice call continuity between old and new systems.

2 Network Topology in LTE

The following schematic provides a good understanding of the overall LTE network infrastructure and elements. The Figure below describes the LTE & UMTS overall network topology, including not only the **EPC** (*Evolved Packet Core*) and **E-UTRAN** (*Evolved UMTS Terrestrial Access Network*), but also other components, in order to show the relationship between them.



Overall network topology in LTE

The new blocks specific to Evolved UMTS evolution are EPC and E-UTRAN.

Other blocks from the classical UMTS architecture are also displayed, such as the **UTRAN** (*the UMTS Access Network*), the PS and the CS Core Networks, respectively, connected to the public (or any private) IP and Telephone Networks. The IMS is located on top of the Packet Core blocks and provide access to both public or private IP networks, and the public telephone network via Media Gateway network entities. The HSS, managing user subscription information is shown as a central node, providing services to all Core Network blocks of 3G and evolved 3G architecture.

2.1 E-UTRAN

History from UMTS

From the first releases of the UMTS standard, the UTRAN architecture was initially very much aligned with 2G/GSM Access Network concepts. The general architecture follows the good old 2G/GSM 'star' model, meaning that a single controller (the **RNC**, *Radio Network Controller*) may possibly control a large number – the typical number in commercial networks is about several hundreds – of radio Base Stations (the NodeB) over the Iub interface. In addition, an inter-RNC Iur interface was defined to allow UTRAN call anchoring at the RNC level and macro-diversity between different NodeB controlled by different RNCs.

The initial UTRAN architecture resulted in a simplified Node B implementation, and a relatively complex, sensitive, high capacity and feature-rich RNC design. In this model, the RNC had to support resource and traffic management features as well as a significant part of the radio protocols.

eNodeB – the Single E-UTRAN Node

Compared with UTRAN, the E-UTRAN structure is quite simple. It is only composed of one network element: the **eNodeB** (*evolved Node B*). The 3G RNC inherited from the 2G **BSC** (*Base Station Controller*) has disappeared from E-UTRAN and the eNodeB is directly connected to the Core Network using the S1 interface. As a consequence, the features supported by the RNC have been distributed between the eNodeB or the Core Network MME or Serving Gateway entities.

eNodeB Functionalities

From a high-level perspective, the new E-UTRAN architecture is actually moving towards WLAN network structures and Wifi or WiMAX Base Stations functional definition.

So functional definition eNodeB (as WLAN access points) support all L1 and L2 features associated to the physical interface, and they are directly connected to network routers. There is no more intermediate controlling node (as the 2G/BSC or 3G/ RNC was). This has the advantage of a simpler network architecture (fewer nodes of different types, which means simplified network operation) and allows better performance over the radio interface.

From a functional perspective, the eNodeB supports a set of legacy features, all related to physical layer procedures for transmission and reception over the radio interface:

- Modulation and de-modulation.
- Channel coding and de-coding.

Besides, the eNodeB includes additional features, coming from the fact that there are no more Base Station controllers in the E-UTRAN architecture. Those features, which are further described in Chapter 4, include the following:

- Radio Resource Control: this relates to the allocation, modification and release of resources for the transmission over the radio interface between the user terminal and the eNodeB.

- Radio Mobility management: this refers to a measurement processing and handover decision.

- Radio interface full L2 protocol: in the OSI, the Layer 2 purpose is to ensure transfer of data between network entities. This implies detection and possibly correction of errors that may occur in the physical layer.

2.2 Evolved Packet Core and its Components

The **EPC** (*Evolved Packet Core*) is composed of several functional entities:

- The **MME** (*Mobility Management Entity*)
- The **HSS** (*Home Subscriber Server*)
- The Serving Gateway.
- The **PDN** Gateway (*Packet Data Network*).
- The **PCRF** (*Policy and Charging Rules Function*) Server.

MME (Mobility Management Entity)

The MME is in charge of all the Control plane functions related to subscriber and session management. From that perspective, the MME supports the following:

- Security procedures – this relates to end-user authentication as well as initiation and negotiation of ciphering and integrity protection algorithms.
- Terminal-to-network session handling – this relates to all the signalling procedures used to set up Packet Data context and negotiate associated parameters like the Quality of Service.
- Idle terminal location management – this relates to the tracking area update process used in order for the network to be able to join terminals in case of incoming sessions.

The MME is linked through the S6 interface to the HSS which supports the database containing all the user subscription information.

HSS (Home Subscriber Server)

The HSS is a central database, and the concatenation of the HLR and the **AuC** (*Authentication Center*) – two functions being already present in pre-IMS 2G/GSM and 3G/UMTS networks. The HLR part of the HSS is in charge of storing and updating when necessary the database containing all the user subscription information, including:

- User identification and addressing – this corresponds to the **IMSI** (*International Mobile Subscriber Identity*) and **MSISDN** (*Mobile Subscriber ISDN Number*) or mobile telephone number.
- User profile information – this includes service subscription states and user-subscribed Quality of Service information (such as maximum allowed bit rate or allowed traffic class).

The HSS may also integrate the authentication center (AUC), which generates the vectors for authentication and security keys. This security information is provided to the HLR and further communicated to other entities in the network. Security information is mainly used for:

- Mutual network-terminal authentication.
- Radio path ciphering and integrity protection, to ensure data and signalling transmitted between the network and the terminal is neither eavesdropped nor altered.

The Serving GW (Serving Gateway)

From a functional perspective, the Serving GW is the termination point of the packet data interface towards E-UTRAN. When terminals move across eNodeB in E-UTRAN, the Serving GW serves as a local mobility anchor, meaning that packets are routed through this point for intra E-UTRAN mobility and mobility with other 3GPP technologies, such as 2G/GSM and 3G/UMTS.

The PDN GW (Packet Data Network Gateway)

Similarly to the Serving GW, the PDN gateway is the termination point of the packet data interface towards the Packet Data Network. As an anchor point for sessions towards the external Packet Data Networks, the PDN GW also supports Policy Enforcement features (which apply operator-defined rules for resource allocation and usage) as well as packet filtering (like deep packet inspection for virus signature detection) and evolved charging support (like per URL charging).

The PCRF (Policy and Charging Rules Function) Server

The PCRF is responsible for policy control decision-making as well as for controlling the flow-based charging functionalities in the Policy Control Enforcement Function (PCEF). The PCRF provides the QoS authorization (QoS class identifier and bit rates) that decides how a certain data flow will be treated in the PCEF and ensures that this is in accordance with the user's subscription profile.

The PCRF Server combines functionalities for the following two UMTS nodes:

- The Policy Decision Function (PDF)
- The Charging Rules Function (CRF)

The PDF is the network entity where the policy decisions are made. As the IMS session is being set up, SIP signalling containing media requirements are exchanged between the terminal and the P-CSCF. At some time in the session establishment process, the PDF receives those requirements from the P-CSCF and makes decisions based on network operator rules, such as:

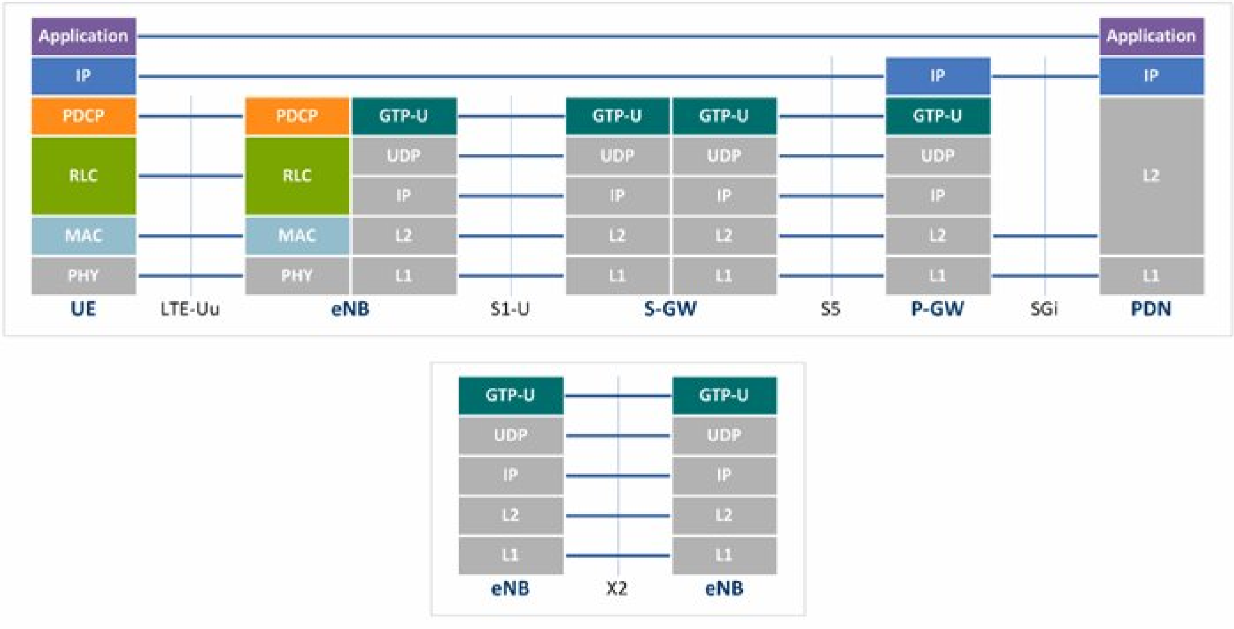
- Allowing or rejecting the media request.
- Using new or existing PDP context for an incoming media request.
- Checking the allocation of new resources against the maximum authorized

The CRFs role is to provide operator-defined charging rules applicable to each service data flow. The CRF selects the relevant charging rules based on information provided by the P-CSCF, such as Application Identifier, Type of Stream (audio, video, etc.), Application Data Rate, etc.

3 Protocol Architecture in LTE

Following figure depicts LTE protocol stack for user and control planes, respectively. The functions of the key layers of the protocol stacks are briefly described below figures.

3.1 LTE User Plane Protocol Stack



LTE user plane protocol stack

PDCP: The PDCP protocol supports efficient transport of IP packets over the radio link. It performs header compression, Access Stratum (AS) security (ciphering and integrity protection) and packet re-ordering/retransmission during handover.

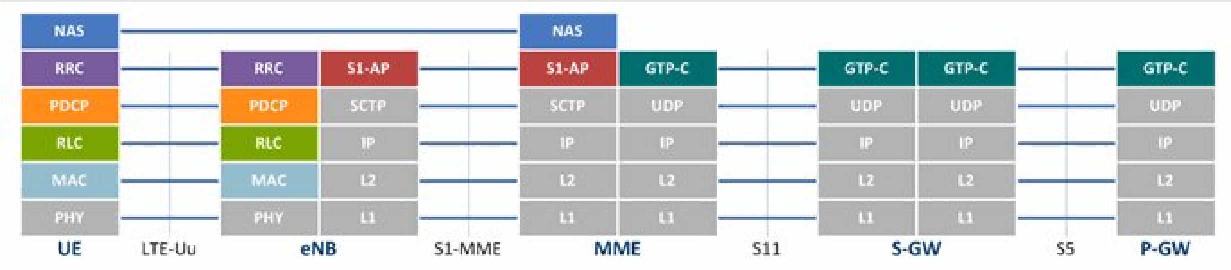
RLC: In the transmitting side, the RLC protocol constructs RLC PDU and provides the RLC PDU to the MAC layer. The RLC protocol performs segmentation/concatenation of PDCP PDUs during construction of the RLC PDU. In the receiving side, the RLC protocol performs reassembly of the RLC PDU to reconstruct the PDCP PDU. The RLC protocol has three operational modes (i.e. transparent mode, acknowledged mode and unacknowledged mode), and each offers different reliability levels. It also performs packet (the RLC PDU) re-ordering and retransmission.

MAC: The MAC layer lies between the RLC layer and PHY layer. It is connected to the RLC layer through logical channels, and to the PHY layer through transport channels. Therefore, the MAC protocol supports multiplexing and de-multiplexing between logical channels and transport channels. Higher layers use different logical channels for different QoS metrics. The MAC protocol supports QoS by scheduling and prioritizing data from logical channels. The eNB scheduler makes sure radio resources are dynamically allocated to UEs and performs QoS control to ensure each bearer is allocated the negotiated QoS.

PHY: The physical layer provides the basic bit transmission functionality over air. It is driven by OFDMA in the downlink and SC-FDMA in the uplink. Physical channels are dynamically mapped to the available resources. Towards higher layers the physical layer offers its data transmission functionality via transport channels. Like in UMTS, a transport channel is a block oriented transmission service with certain characteristics regarding bit rates, delay, collision risk and reliability. In contrast to 3G WCDMA or even 2G GSM there are no dedicated transport or physical channels anymore, as all resource mapping is dynamically driven by the cheduler.

GTP-U: GTP-U protocol is used to forward user IP packets over S1-U, S5 and X2 interfaces. When a GTP tunnel is established for data forwarding during LTE handover, an End Marker packet is transferred as the last packet over the GTP tunnel.

3.2 LTE Control Plane Protocol Stack



LTE control plane protocol stack

NAS: NAS protocol performs mobility management and bearer management functions.

RRC: RRC protocol supports the transfer of the NAS signaling. It also performs functions required for efficient management of the radio resources. The main functions are as follows:

- Broadcasting of system information
- Setup, reconfiguration, reestablishment and release of the RRC connection
- Setup, modification and release of the radio bearer

X2AP: X2AP protocol supports UE mobility and SON functions within the E-UTRAN. To support UE mobility, the X2AP protocol provides functions such as user data forwarding, transfer of SN status and UE context release. For SON functions, eNBs exchange resource status information, traffic load information and eNB configuration update information, and coordinate each other to adjust mobility parameters using the X2AP protocol.

S1AP: S1AP protocol supports functions such as S1 interface management, E-RAB management, NAS signaling transport and UE context management. It delivers the initial UE context to the eNB to setup E-RAB(s) and manages modification or release of the UE context thereafter.

GTP-C: GTP-C protocol supports exchange of control information for creation, modification and termination for GTP tunnels. It creates data forwarding tunnels in case of LTE handover.

4 Physical and Logical Channels in LTE

As for most radio communication systems, the radio interface of E-UTRAN faces many challenges. In terms of requirements, the E-UTRAN shall be able to transmit high-rate and low-latency information in the most efficient way. However, not all the information flows require the same protection against transmission errors or Quality of Service handling.

In general, it is critical, especially in the case of radio mobility, that E-UTRAN signalling messages are transmitted as fast as possible, usingthe best error-protectionscheme. On the other hand, voice or data streaming applications can accept a reasonable frame loss due to radio transmission. Interactive connection-oriented applications (such as Web browsing) are also different, as the end-to-end retransmission can help to recover from radio propagation issues

In order to be flexible and allow different schemes for data transmission, the E-UTRAN specifications introduce several types of channels:

- The logical channels – what is transmitted.
- The transport channels – how it is transmitted.
- The physical channels

The **logical channels** correspond to data-transfer services offered by the radio interface protocols to upper layers. Basically, there are only two types of logical channels: the control channels (for the transfer of Control plane information) and the traffic channels (for the transfer of User plane information). Each of the channels of these two categories corresponds to a certain type of information flow.

The **transport channels** describe how and with what characteristics data are transferred over the radio interface. For example, the transport channels describe how the data are protected against transmission errors, the type of channel coding, CRC protection or interleaving which is being used, the size of data packets sent over the radio interface, etc.

As in the specification, the transport channels are classified into two categories:

- the downlink transport channels (from the network to the terminal) and
- the uplink transport channels (from the terminal to the network).

The **physical channels** are the actual implementation of the transport channel over the radio interface. They are only known to the physical layer of E-UTRAN and their structure is tightly dependent on physical interface OFDM characteristics.

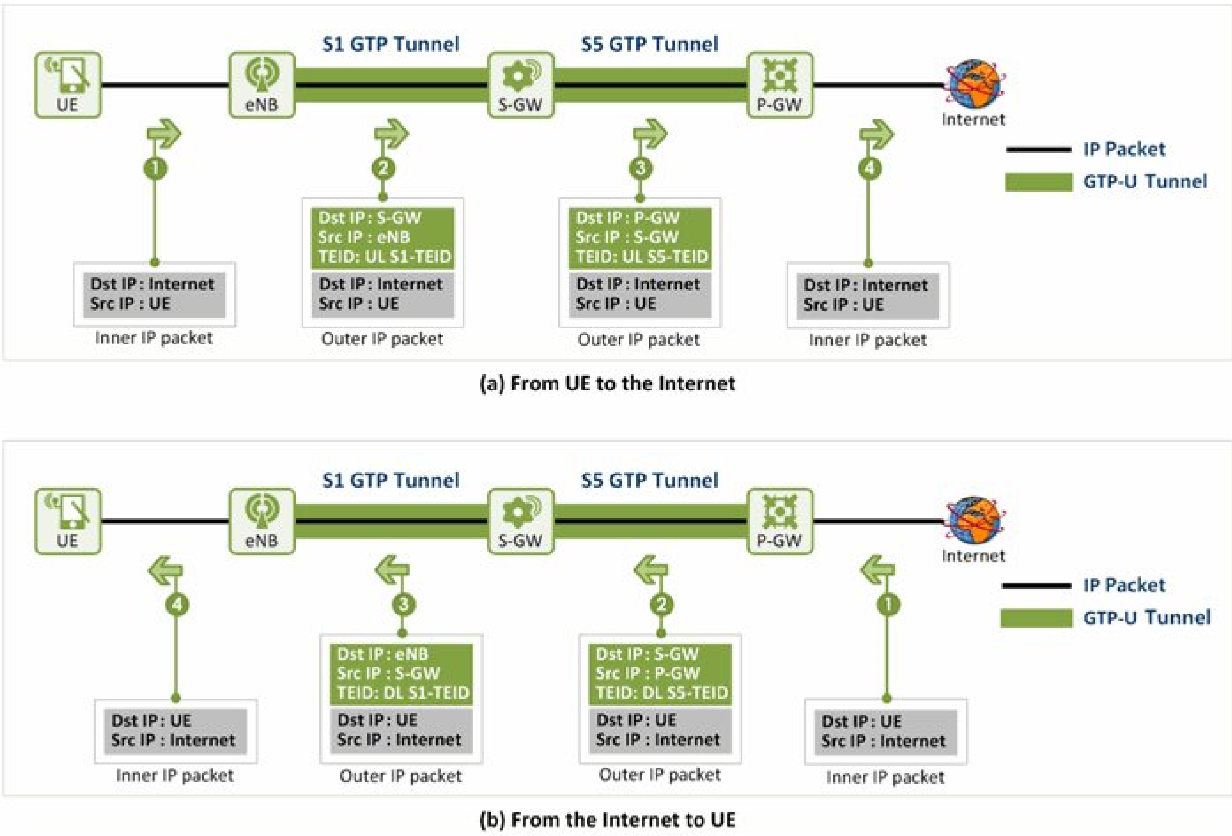
5 Traffic Flow in the LTE Network

Next figure shows the flow of user plane traffic accessing the Internet in the LTE network reference architecture.

IP packets are forwarded through the GTP tunnel over S1-U and S5 interfaces. These GTP tunnels are established per EPS bearer when a user is attached to the LTE network.

More than one EPS bearer is established on each of the S1-U and S5 interfaces. So, in order to identify these bearers, a *Tunnel Endpoint Identifier* (**TEID**) is assigned to the end points (UL and DL) of each GTP tunnel (When identifying a GTP tunnel, a TEID, IP address and UDP port number are used in general.

Here, however, for convenience of description, only a TEID is used for this purpose. The receiving end side of the GTP tunnel locally assigns the TEID value the transmitting side has to use. The TEID values are exchanged between tunnel endpoints using control plane protocols.



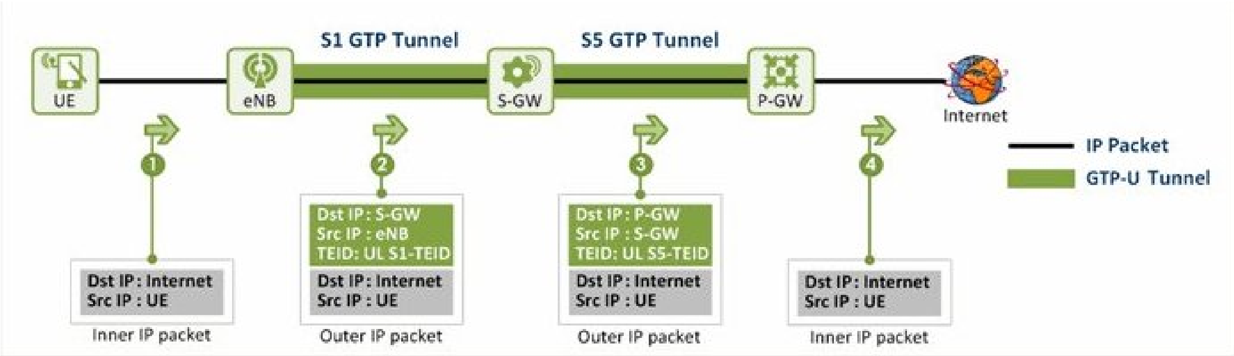
Traffic flow in the LTE network

When a GTP tunnel is established on the S1-U interface, the S-GW assigns a TEID (UL S1-TEID in figure (a) for uplink traffic and the eNB assigns a TEID (DL S1-TEID in figure (b)) for downlink traffic. The TEID values of the S1 GTP tunnel are exchanged between the eNB and the S-GW using S1AP and GTP-C messages.

Likewise when a GTP tunnel is established on the S5 interface, the P-GW assigns a TEID (UL S5-TEID in figure (a)) for uplink traffic and the S-GW assigns a TEID (DL S5-TEID in figure (b)) for downlink traffic. The TEID values of the S5 GTP tunnel are exchanged between the S-GW and the P-GW using GTP-C protocol.

When a user IP packet is delivered through a GTP tunnel on the S1-U and S5 interfaces, the eNB, S-GW and P-GW forward the user IP packet by encapsulating with the TEID assigned by the receiving peer GTP entity. In uplink direction, the S-GW builds a one-to-one mapping between an S1 GTP tunnel (UL S1-TEID) and an S5 GTP tunnel (UL S5-TEID) to terminate the S1 GTP tunnel and forward the user IP packet into the S5 GTP tunnel.

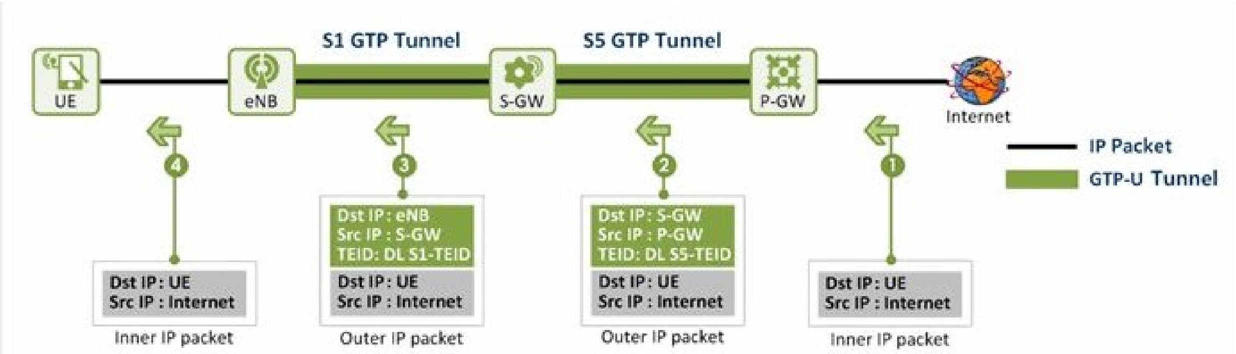
5.1 Traffic flow in uplink direction



Traffic flow from UE to Internet

1. A UE transfers user IP packets to an eNB over LTE-Uu interface.
2. The eNB encapsulates the user IP packets with the S1 GTP tunnel header and forwards the resulting outer IP packets to the S-GW. Here, the eNB selected a "TEID" value (i.e. UL S1-TEID), "Destination IP Address" (i.e. S-GW IP address), and "Source IP Address" (i.e. eNB IP address) to make the S1 GTP tunnel header.
3. After receiving the outer IP packets, the S-GW strips off the S1 GTP tunnel header, encapsulates the user IP packets (the inner IP packets) with the S5 GTP tunnel header and forwards the resulting outer IP packets to the P-GW. Here the S-GW selected a "TEID" value (i.e. UL S5-TEID), "Destination IP Address" (i.e. P-GW IP address), and "Source IP Address" (i.e. S-GW IP address) to make the S5 GTP tunnel header.
4. After receiving the outer IP packets, the P-GW gets the user IP packets by stripping off the S5 GTP tunnel header and transfers them to the Internet through IP routing.

5.2 Traffic flow in downlink direction



Traffic flow from Internet to UE

1. A P-GW receives IP packets destined for a UE over the Internet.
2. The P-GW encapsulates the user IP packets with the S5 GTP tunnel header and forwards the resulting outer IP packets to the S-GW. Here, the P-GW selected a "TEID" value (i.e. DL S5-TEID), "Destination IP Address" (i.e. S-GW IP address), and "Source IP Address" (i.e. P-GW IP address) to make the S5 GTP tunnel header.
3. After receiving the outer IP packets, the S-GW strips off the S5 GTP tunnel header, encapsulates the user IP packets (the inner IP packets) with the S1 GTP tunnel header and forwards the resulting outer IP packets to the eNB. Here, the S-GW selected a "TEID" value (i.e. DL S1-TEID), "Destination IP Address" (i.e. eNB IP address), and "Source IP Address" (i.e. S-GW IP address) to make the S1 GTP tunnel header.
4. After receiving the outer IP packets, the eNB gets the user IP packets by stripping off the S1 GTP tunnel header and transfers them to the UE through the Data Radio Bearer (DRB) over the radio link.

6 Voice Transport in LTE

The LTE standard supports only packet switching technology with its all-IP network. The missing circuit-switched domain provides some challenges to deliver voice via an LTE network.

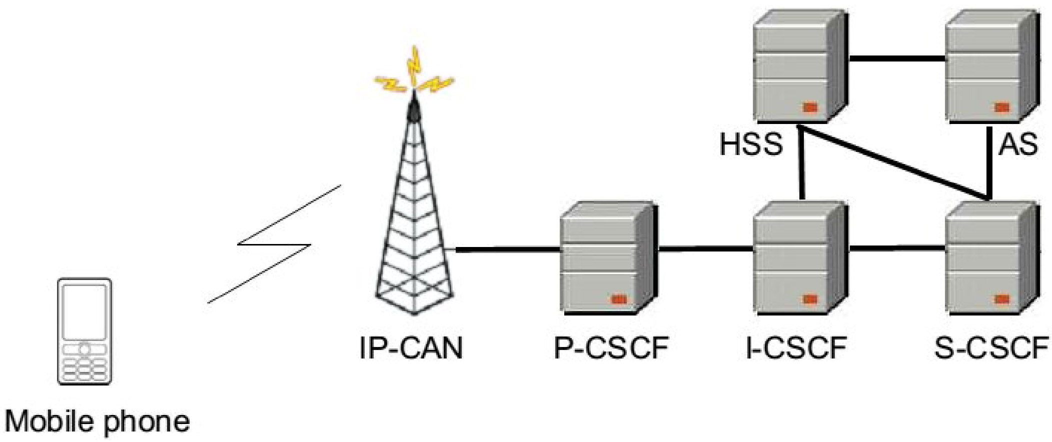
Hence voice calls in GSM, UMTS and CDMA2000 are circuit-switched, so with the adoption of LTE, carriers will have to re-engineer their voice call network. Three different approaches sprang up:

- Voice over LTE (VoLTE)
- Circuit-switched fallback (CSFB)
- Simultaneous voice and LTE (SVLTE)

6.1 Voice over LTE

Voice communication is in LTE natively supported only using IMS services, with specific profiles for control and media planes.

IMS is an access-independent overlay to existing network architectures, guaranteeing seamless service continuity, not only for voice, but also e.g. for video application. The first version of IMS was standardized in 3GPP release 5, with many enhancements specified in subsequent releases. IMS needs to be implemented on both the network as well as the device side, whereas rollout of IMS in commercial networks was slower than originally expected.



IMS subsystem in LTE

For LTE, the **IP-CAN IP** (*Connectivity Access Network*) would be composed of the EPS and the E-UTRAN.

The call session control functions are the core components of the IMS. There are three CSCF:

- **P-CSCF** (*Proxy Call State Control Function*): The P-CSCF is the user to network proxy. In this respect all SIP signalling to and from the user runs via the P-CSCF whether in the home or a visited network.

- **I-CSCF** (*Interrogating Call State Control Function*): The I-CSCF is used for forwarding an initial SIP request to the S-CSCF. When the initiator does not know which S-CSCF should receive the request.

- **S-CSCF** (*Serving Call State Control Function*): The S-CSCF undertakes a variety of actions within the overall system, and it has a number of interfaces to enable it to communicate with other entities within the overall system.

The **HSS** (*Home Subscriber Server* is the main subscriber database used within IMS. The IMS HSS provides details of the subscribers to the other entities within the IMS network, enabling users to be granted access or not dependent upon their status.

The **AS** (*Application Server*) provides specific IP applications such as messaging.

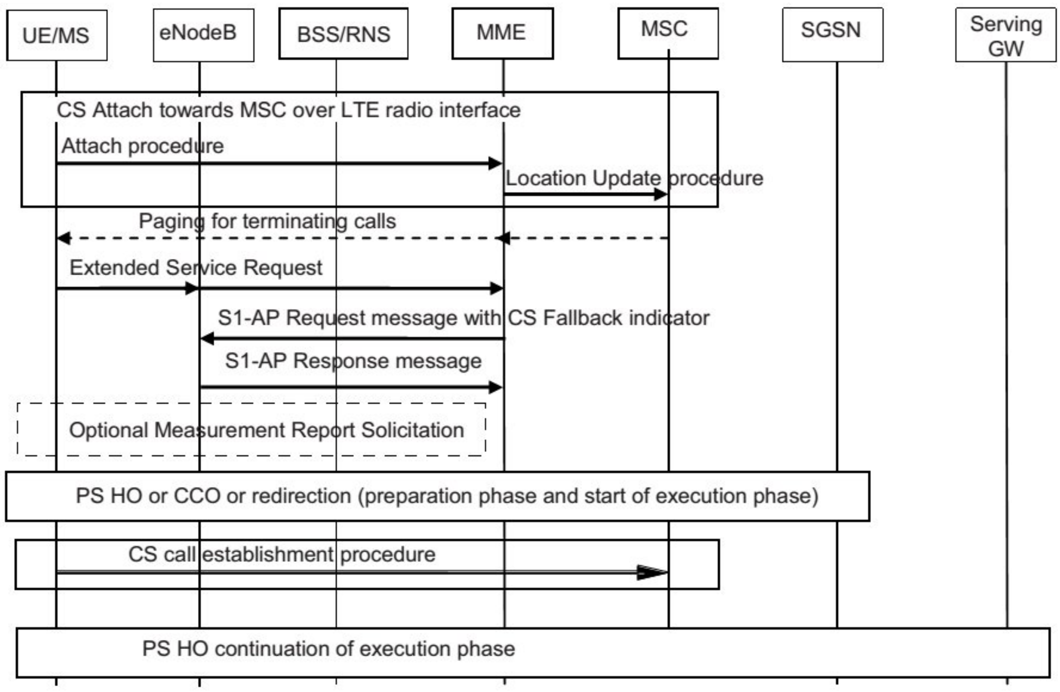
The IMS calls for VoLTE are processed by the subscriber's S-CSCF in the home network. The connection to the S-CSCF is via the P-CSCF. Dependent upon the network in use and overall location within a network, the P-CSCF will vary, and a key element in the enablement of voice calling capability is the discovery of the P-CSCF.

The continuity of voice calls needs to be guaranteed by handover to a legacy technology such as GSM. This is achieved by a feature called **SRVCC** (*Single Radio Voice Call Continuity*).

6.2 CSFB – Circuit-Switched FallBack

In case IMS services are not deployed from the start and/or LTE provides only data services, when a voice call is to be initiated or received, it will fall back to the circuit-switched domain by the **CSFB** (*Circuit-Switched FallBack*) mechanism. CSFB allows circuit-switched voice calls to be handled via legacy radio access technology for terminals that are camped on LTE.

When using this solution, operators just need to upgrade the MSC instead of deploying the IMS. However, the disadvantage is longer call setup delay.



Message sequence diagram for CSFB from LTE to UMTS

Figure shows the message flow for a CSFB call from LTE to UMTS, including paging from the MSC via the SGs interface and MME in the case of UE-terminated calls, and the sending of an Extended Service Request message from the UE to the MME to trigger either a handover or redirection to the target radio access technology.

6.3 Simultaneous voice and LTE (SVLTE)

In this approach, the handset works simultaneously in the LTE and circuit switched modes, with the LTE mode providing data services and the circuit switched mode providing the voice service.

This is a solution solely based on the handset, which does not have special requirements on the network and does not require the deployment of IMS either. The disadvantage of this solution is that the phone can become expensive with high power consumption.

One additional approach which is not initiated by operators is the usage of over-the-top content (OTT) services, using applications like Skype and Google Talk to provide LTE voice service.

7 Quality of Services in LTE

In a typical case, multiple applications may be running in a UE at the same time, each one having different QoS requirements. For example, an UE can be engaged in a VoIP call while at the same time browsing a web page or downloading an FTP file. VoIP has more stringent requirements for QoS in terms of delay and delay jitter than web browsing and FTP, while the latter requires a much lower packet loss rate. In order to support multiple QoS requirements, different bearers are set up within EPS, each being associated with a QoS.

Broadly, bearers can be classified into two categories based on the nature of the QoS they provide:

- The **GBR** (*Guaranteed Bit Rate*) bearers which can be used for applications such as VoIP. These have an associated GBR value for which dedicated transmission resources are permanently allocated (e.g. by an admission control function in the eNodeB) at bearer establishment/modification. Bit rates higher than the GBR may be allowed for a GBR bearer if resources are available. In such cases, a *Maximum Bit Rate* (**MBR**) parameter, which can also be associated with a GBR bearer, sets an upper limit on the bit rate which can be expected from a GBR bearer.

- Non-GBR bearers which do not guarantee any particular bit rate. These can be used for applications such as web browsing or FTP transfer. For these bearers, no bandwidth resources are allocated permanently to the bearer.

In the access network, it is the eNodeB's responsibility to ensure that the necessary QoS for a bearer over the radio interface is met. Each bearer has an associated *Class Identifier* (**QCI**), and an *Allocation and Retention Priority* (**ARP**).

Each QCI is characterized by priority, packet delay budget and acceptable packet loss rate. The QCI label for a bearer determines the way it is handled in the eNodeB. Only a dozen such QCIs have been standardized so that vendors can all have the same understanding of the underlying service characteristics and thus provide the corresponding treatment, including queue management, conditioning and policing strategy. This ensures that an LTE operator can expect uniform traffic handling behaviour throughout the network regardless of the manufacturers of the eNodeB equipment. The set of standardized QCIs and their characteristics (from which the PCRF in an EPS can select) is provided in Table below.

Standardized QoS Class Identifiers (QCIs) for LTE

QCI	Resource type	Priority	Packet delay budget (ms)	Packet error loss rate	Example services
1	GBR	2	100	10−2	Conversational voice
2	GBR	4	150	10−3	Conversational video (live streaming)
3	GBR	5	300	10−6	Non-conversational video (buffered streaming)
4	GBR	3	50	10−3	Real time gaming
5	Non-GBR	1	100	10−6	IMS signalling
6	Non-GBR	7	100	10−3	Voice, video (live streaming), interactive gaming
7	Non-GBR	6	300	10−6	Video (buffered streaming)
8	Non-GBR	8	300	10−6	TCP-based (e.g. WWW, e-mail) chat, FTP, p2p file sharing, progressive video, etc.
9	Non-GBR	9	300	10−6	

The priority and packet delay budget (and, to some extent, the acceptable packet loss rate) from the QCI label determine the RLC mode configuration, and how the scheduler in the MAC handles packets sent over the bearer (e.g. in terms of scheduling policy, queue management policy and rate shaping policy). For example, a packet with a higher priority can be expected to be scheduled before a packet with lower priority.

8 Advancement in LTE

With the completion of LTE Release 8, 3GPP started to look into ways to further evolve LTE for the future, in order to build upon the existing LTE technology and to ensure that LTE remains the leading global standard for mobile broadband.

Enhanced performance can in principle be achieved in two ways – by using more radio spectrum, and by using the available spectrum more efficiently.

The main components of LTE-Advanced that are added to LTE in Release 10 are:

- Carrier aggregation;
- Enhanced downlink multiple antenna transmission;
- Uplink multiple antenna transmission;
- Relaying;
- Support for heterogeneous network deployments.

Data rates of the order of 1 Gbps might theoretically be achieved using contiguous bandwidths of 40 MHz or more. However, competition for spectrum and fragmentation of the available spectrum makes it unrealistic to expect such large contiguous bandwidths in most cases. LTE-Advanced therefore makes use of carrier aggregation to support such large bandwidths. This also has the advantages of limiting the cost of equipment and enabling much of the technology developed for LTE Release 8 to be reused. Each 'component carrier' within an aggregation is designed to be fundamentally similar to an LTE Release 8 carrier so that they can be configured in a backward-compatible way and used by legacy UEs if desired. Up to five component carriers with a bandwidth of up to 20 MHz each can be aggregated in LTE-Advanced to make efficient use of the available spectrum and achieve the desired total bandwidth and peak data rate.

LTE-Advanced can also make use of carrier aggregation to support deployments of heterogeneous networks consisting of a layer of macrocells and a layer of small cells coexisting with at least one carrier being common between them. In such a deployment, transmissions from one cell can interfere strongly with the control channels of another, thus impeding scheduling and signalling. LTE-Advanced supports cross-carrier scheduling to enable control signalling to be transmitted on one component carrier corresponding to data transmissions on another; in this way, control channel interference between macrocells and small cells can be avoided.

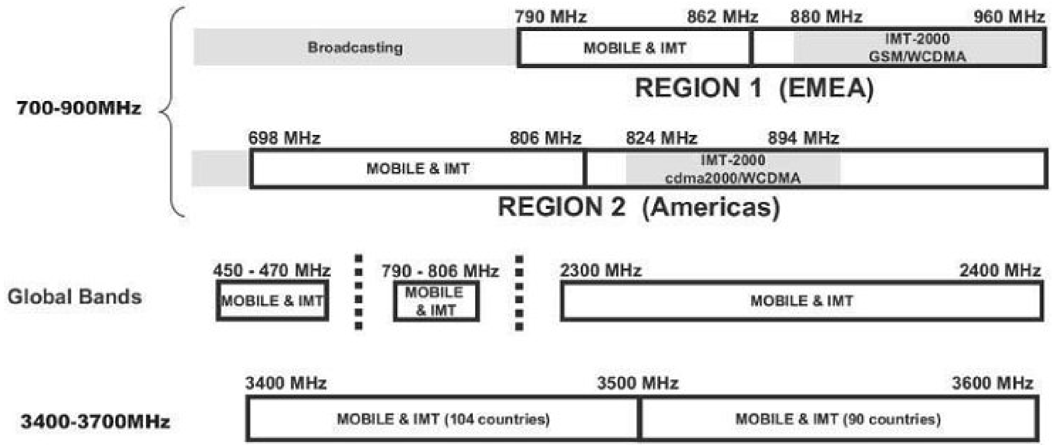
The existence of internationally identified common frequency bands is a key factor for significant economies of scale in the development and production of terminals. A key outcome of the WRC-2007 (World Radiocommunication Conference, held in Geneva in 2007), was that a total of 136 MHz of new global spectrum was allocated for use by International Mobile Telecommunication-designated radio technologies:

- 450–470 MHz;

- 790–806 MHz;
- 2300–2400 MHz.

Other region-specific bands were also allocated:

- 790–862 MHz for ITU Region 1 (EMEA) and ITU Region 3 (all other Asia Pacific);
- 698–806 MHz for ITU Region 2 (North and South America) and ITU Region 3 (nine countries, including Japan, China and India);
- 3400–3600 MHz allocated to mobile use on a primary basis for ITU Region 1 (EMEA in 82 countries), ITU Region 2 (Americas in 14 countries, except US/Canada) and Region 3.



Allocation of new global spectrum resulting from WRC-2007.

All new bands identified by the WRC 2007 are valid generically for International Mobile Telecommunication technologies.