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VYSOKÉ UČENÍ TECHNICKÉ V BRNĚ

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A KOMUNIKAČNÍCH TECHNOLOGIÍ

DEPARTMENT OF TELECOMMUNICATIONS

ÚSTAV TELEKOMUNIKACÍ

VOLTE SERVICE IMPLEMENTATION IN EPS-IMS NETWORKS

IMPLEMENTACE SLUŽBY VOLTE DO SÍTÍ EPS-IMS

MASTER'S THESIS

DIPLOMOVÁ PRÁCE

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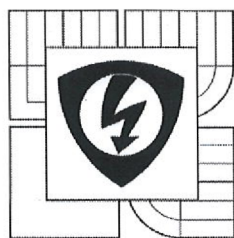AUTOR PRÁCE

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VYSOKÉ UČENÍ TECHNICKÉ V BRNĚ

**Fakulta elektrotechniky
a komunikačních technologií**

Ústav telekomunikací

Diplomová práce

magisterský navazující studijní obor
Telekomunikační a informační technika

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Ročník: 2   *Akademický rok:* 2015/16

NÁZEV TÉMATU:

Implementace služby VoLTE do sítí EPS-IMS

POKYNY PRO VYPRACOVÁNÍ:

Seznamte se s mobilními sítěmi EPS a subsystémem IMS. Zaměřte se na hlasovou službu, rozeberte v současnosti používaná řešení a porovnejte je. Detailněji prostudujte problematiku nasazení služby VoLTE a v závislosti na možnostech Ústavu telekomunikací a její experimentální sítě EPS-IMS popište konkrétní nasazení VoLTE, realizujte několik VoLTE relací za pohybu uživatele s cílem dosáhnout procedury handover, zachyťte provoz a analyzujte ho. Na základě nabytých zkušeností navrhněte laboratorní úlohu pro předmět MKPM.

DOPORUČENÁ LITERATURA:

[1] GUNNAR H. SAE / EPC from A-Z. Inacon, ISBN 978-3-936273-59-5, Germany, 2011
[2] LESCUYER, P., LUCIDARME, T. Evolved Packet System: The LTE and SAE Evolution of 3G UMTS. John Wiley & Sons, ISBN: 978-0-470-05976-0, GB, 2008

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Vedoucí práce: doc. Ing. Vít Novotný, Ph.D.
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předseda oborové rady

ABSTRAKT

Diplomová práce popisuje VoLTE službu, vývoj a nasazení LTE (zaváděcí fázi, skutečný LTE stav a výhledy do budoucna atd.), EPC-IMS architekturu (popis funkce uzlu, rozhraní atd.) Komunikace mezi uzly a funkce, rozhraní a protokoly jsou používány v průběhu signalizace (SIP SDP) a datový tok (RTCP RTP). Práce stručně popisuje základní toky hovorů, typy nosičů (GBR and N-GBR), a to vytvoření / mazaní nosičů během komunikace. Další část diplomové práce o implementaci volte, instalace a konfigurace IMS. Závěrečná část diplomové práce popisuje zkoušky sítě a, analýzu protocolu.

KLÍČOVÁ SLOVA

VoLTE, Call flow, IMS, EPC, CSCF, HSS, mobile network, SIP, SDP, RTCP, RTP

ABSTRACT

The master's thesis describes VoLTE service, LTE evolution and deployment (deployment phases, actual LTE state and future perspectives etc.), EPC-IMS architecture (functional node description, interfaces etc.). Communications between nodes and functions, interfaces and protocols which are used during signaling (SIP-SDP) and data flow (RTCP RTP). The thesis briefly describes basic call flows, bearers types (GBR and N-GBR) and their establishment/delete during communication. The next part of master's thesis is about VoLTE implementation solutions, IMS installation and configuration. The final part of the master's thesis describes the network and protocols tests and analyzes.

KEYWORDS

VoLTE, Call flow, IMS, EPC, CSCF, HSS, mobile network, SIP, SDP, RTCP, RTP

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Prohlášení

Prohlašuji, že svou diplomovou práci na téma „IMPLEMENTACE SLUŽBY VOLTE DO SÍTÍ EPS-IMS" jsem vypracoval samostatně pod vedením vedoucího diplomové práce a s použitím odborné literatury a dalších informačních zdrojů, které jsou všechny citovány v práci a uvedeny v seznamu literatury na konci práce.

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V Brně dne …………….   ……………………………………………

                                            podpis autora

Poděkování

Děkuji vedoucímu práce doc. Ing. Vítu Novotnému, Ph.D za velmi užitečnou metodickou pomoc a cenné rady při zpracování diplomové práce.

V Brně dne ……………..   …………………………………….……

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Contents

1 Introduction

This work is about the hottest voice service in LTE network – VoLTE. Implementing VoLTE services for public mobile network operators is a key moment to aggregate all signaling and data rates under an IP network. This solution opens new IP-based opportunities for mobile network services, and approaches the biggest technology in telephony history – Circuit Switched Networks.

It would be almost impossible to explain everything about VoLTE in a single work without making it a thick book and I aim on my tasks of solving the real problems that I met during work.

In the second chapter, I analyze LTE architecture and study the solutions developed by vendors, description of 3GPP, GSMA Technical Subscription and implementation guidelines to design the solution that corresponds to the thesis task requirements. I describe Evolved Packet Core, the main IMS nodes, functions and interfaces.

In the third chapter, I describe the LTE evolution, deployment and goals that allowed implementing voice service to 4G network. Currently, the FEKT VUT LTE network is in phase 1 of the implementation, it supports only data traffic, no CSFB or VoLTE voice service implement. One of the tasks of this work is solve this task implement the voice services in the faculty LTE network.

The forth part is about VoLTE solution at the faculty LTE network and its configuration. This part was written based on the materials of VimpelCom workshop on Huawei EPC devices and it configuration for support VoLTE service that took place in Ekaterinburg, Russia on June, 27-29. Unfortunately, it was an internal course and the most of materials was for internal use only, but using my experience I made configuration templates for the faculty LTE network topology.

In addition, I performed RAN tests and traffic capture to analyze and describe the network principles.

The fifth part of the work is devoted to IMS configuration. I installed and configured free open source Kamailio IMS (ex SER) with Fraunhofer HSS and Sip Express Media Server (SEMS) Application Server (AS).

In the sixth chapter, I described the Laboratory Exercise that was created in frame of this work.

At last, I conclude the paper with a brief discussion of the solved tasks.

2 VoLTE architecture

The VoLTE logical architecture is based on the 3GPP defined architecture and principles and compared with GSMA principles, vendor's devices and Mobile Network Providers implementation like AT&T and VimpelCom. The main functional nodes of the VoLTE architecture are described below.

2.1 UE functions

The VoLTE user equipment was use to connect eNB via the LTE-Uu radio interface.

For the establishment of a RRC connection RRC CONNECTIONREQUEST message was used. This message was sent as part of the Random Access procedure and transferred using SRB0 on the Common Control Channel (CCCH).

After establish the RRC connection, UE initiates the Attach Request to the eNodeB. Attach Request include the next information about UE:

- EPS Attach Type
- NAS key set identifier
- IMSI
- UE network capability
- DRX parameters
- PDN Type (set to IPv4v6)
- PCO (P-CSCF IPv4 Address Request, P-CSCF IPv6 Address Request, IPv4 Link MTU Request)
- Voice Domain Preference and UE's Usage Setting (indicating support of IMS voice)
- ESM message container

To interact between UEs and network elements with created bearers, using Mb Interface (Media Bearer). The protocol is based on symmetric RTP/RTCP over UDP.

Ut Interface is between the UE and the TAS and allows user configuration of the supplementary services specified for VoLTE service. The protocol used on the Ut interface is XCAP.

2.2 E-UTRAN architecture

eNodeB is element of radio access network (E-UTRAN), handles the radio communications between the mobile and the evolved packet core and it is just only one component in E-UTRAN. E-UTRAN based on the "Flat architecture", that allow to eliminates centralized controllers like RNC or BSC, and consist of a single node, the eNodeBs (eNB).
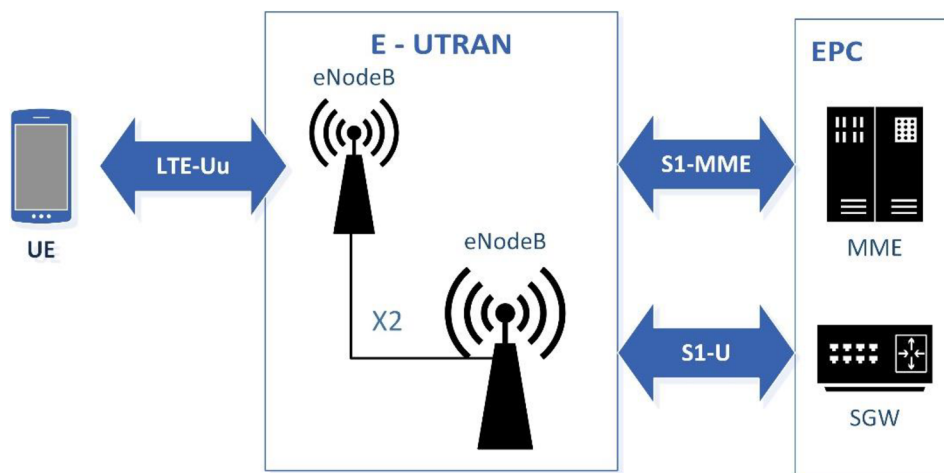
Picture 2-1 E-UTRAN Architecture

For pooling resources, each eNB was connect to multiple MMEs and SGWs, that covering same geographical area. All–IP design – allow to delivering all services over the same infrastructure.

S1 interface connects E-UTRAN to EPC, exist two kinds of this interface:

S1u – user plane: using GPRS Tunneling Protocol (GTP) on UDP. May be only one GTP tunnel per radio bearer. For QoS assigns level of service and priority to each packet used Differentiated Services Code Point (DSCP).

S1mme – Control plane interface between EUTRAN and MME. Carries S1 Application Protocol (S1AP) messages. Handle EPS bearer (setup and release), paging, NAS and handover
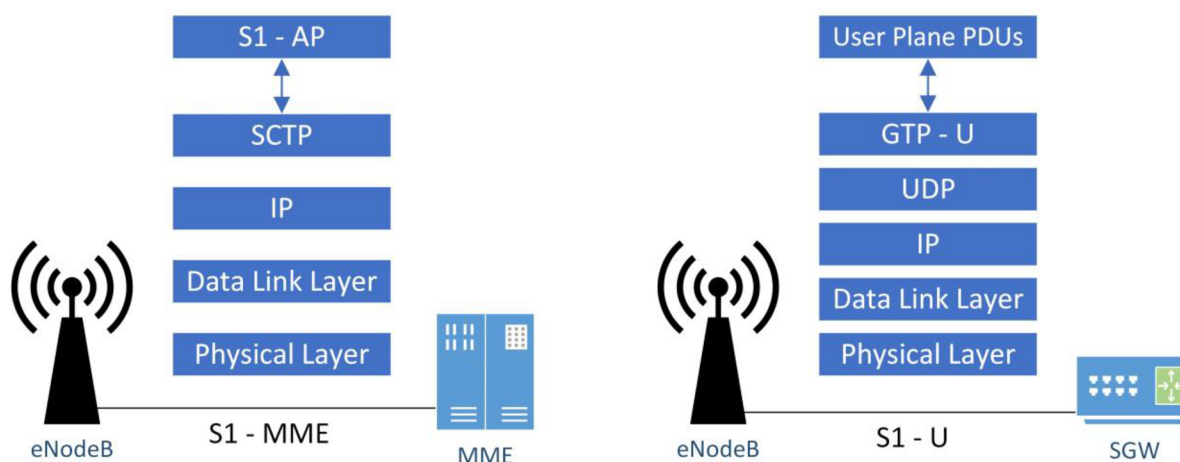
Picture 2-2 S1 Interfaces

signaling. For guaranteed data delivery over IP using Stream Control Transmission Protocol (SCTP), each SCTP association can support multiple UE.

X2 interfaces were the new feature of LTE E-UTRAN. This interface allows to connects eNodeBs directly to separate into E-UTRAN signaling and user traffic for supporting intra LTE mobility. Provides redundancy and load sharing. eNodeBs interconnected via X2 interface. If X2 interface available between two neighbor eNodeBs, this interface managing to handover within Mobile Management Entity (MME). Similar to S1u use GTP to carry user traffic. X2-C interface using for signaling and X2-U interface using for user plane. X2AP protocol create tunnels between eNodeBs during inter-eNodeB mobility.

The main functions of eNodeB below:

- Radio Resource Management (RRM) include radio bearer control and radio admission control
- IP header compression and user data encryption
- Uplink and downlink resource scheduling
- Transmission of paging messages and broadcast information
- Selection of MME: After getting Attach request from UE, the eNodeB select the MME
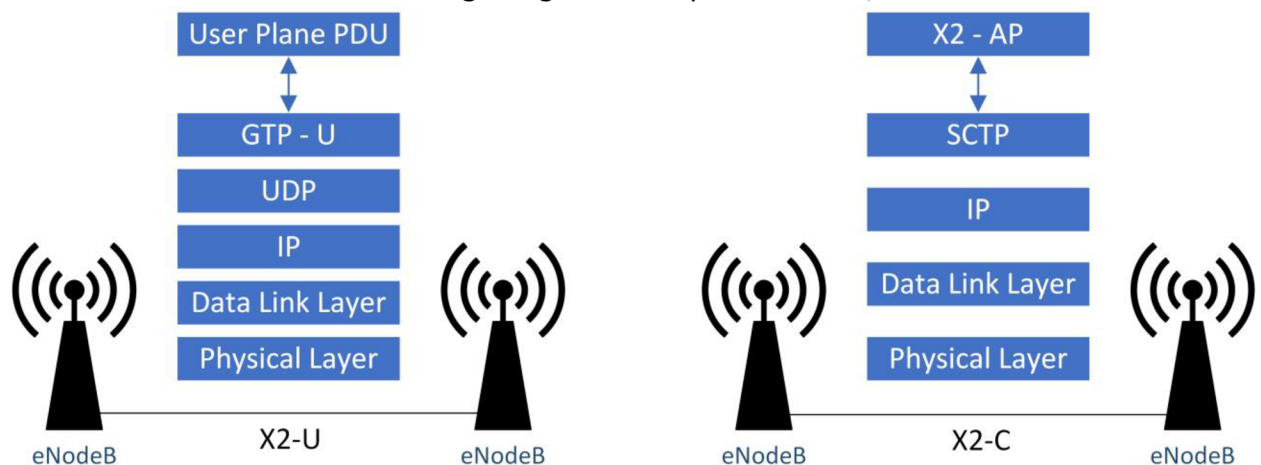
Picture 2-3 X2 Interfaces

from the RRC parameters and forward the Attach request to the MME after add cell information, TAI+ECGI location information and Selected Network.
- Measurements (quality, performance)
- Routing of user data to the S-GW

LTE radio access network RAN are based on OFDMA orthogonal frequency-division multiple access and MIMO (multiple-input and multiple-output) technologies. OFDMA goals is:

- Scalability – able to utilize variable bandwidth.
- Time and frequency scheduling – radio resources can be allocated over multiple channels and/or multiple transmission symbols.
- Reduced interference – channels don't interfere.
- Higher data rates – OFDM have plenty of channels and user can get multiple assigned.
- MIMO support.
- In a difference of FDM, OFDM have not guard bands, which save bandwidth.

2.3 Evolved Packet Core (EPC) architecture

2.3.1 Functions and nodes

This part gives an overview of Evolved Packet Core network its components and interfaces and describe the main (based on FEKT LTE network) nodes, interfaces and protocols.
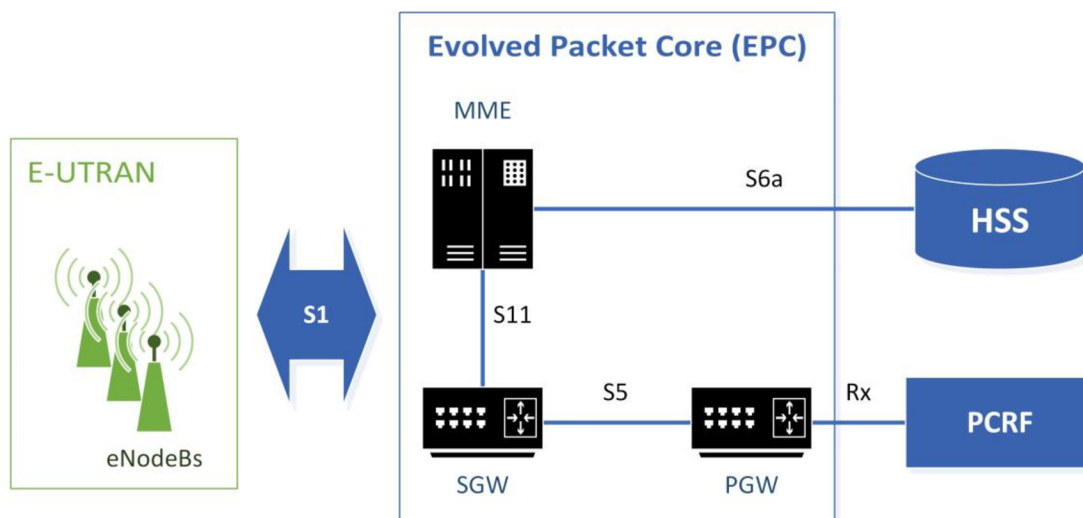
Picture 2-4 EPC Architecture

MME – Mobility Management Entity is the key control-node for the LTE access network. It is signaling node only – not involved in user traffic. MME responsible for functions related to UE mobility:

- Managing and storing UE contexts
- Generation and allocation of temporary UE IDs
- Idle state mobility and roaming
- Security functions (authentication, authorization)
- Bearer path control (S-GW and P-GW selection)
- MME provides the control plane function for mobility between LTE and 2G/3G access networks and interfaces with the home HSS for roaming UEs.
- Support for other call control and session management entities

In the largest mobile network MME was pooled for support redundancy, between MMEs, that serving the same area using load balancing. The **S10** interface provides for MME – MME information transfer and was use to enable MME relocation. The protocol used on the S10 interface is GPRS Tunneling Protocol-Control plane (GTPv2-C). For authenticating and authorizing user access to HSS **S6a** interface was used, for DB requests in practice using Authentication Authorization and Accounting (AAA) service based on Diameter protocol. The S11 interface is between the MME and S-GW to support mobility and bearer management. The protocol used on the S11 interface is GPRS Tunneling Protocol-Control plane (GTPv2-C). (2)

The Non-Access Stratum is a set of protocols in the Evolved Packet System. The NAS is used to transfer non-radio signaling between the User Equipment (UE) and the Mobility Management Entity (MME) for an LTE/E-UTRAN access. (4) NAS register UE in MME, initiate and maintaining EPS bearers and moving S1 and radio bearers during mobility. Carried over LTE-Uu and S1mme interfaces.
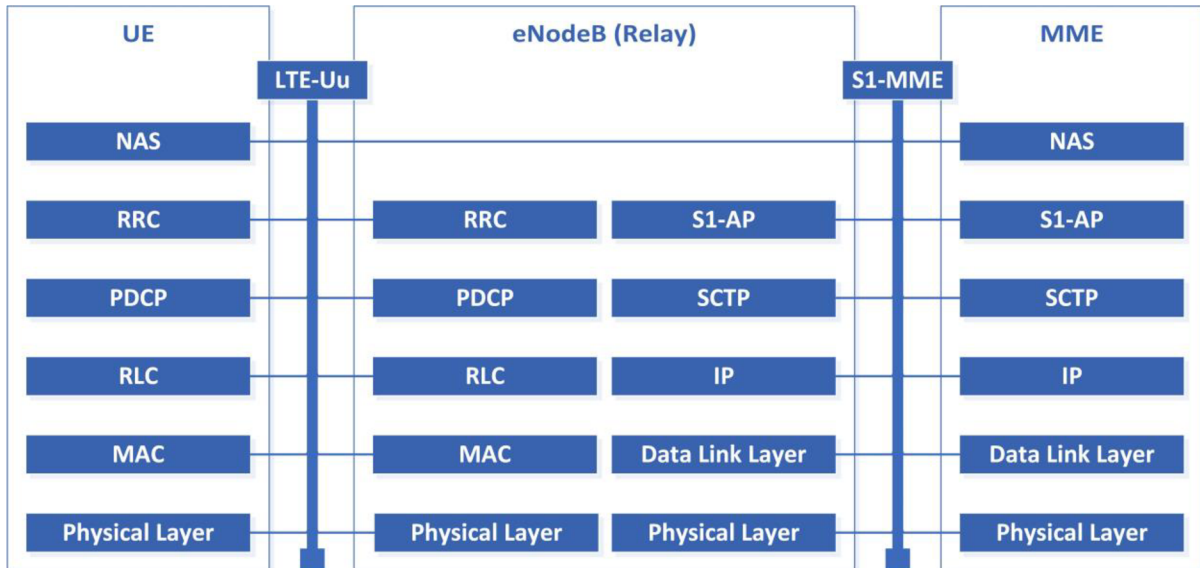
Picture 2-5 Non-Access Stratum communication

Home Subscriber Server (HSS) – network master database, which holds user description data (ID, numbering and service profiles) and provide authentication and authorization to the MME and IMS core during UE attach and IMS registration. For DB requests in practice using Authentication Authorization and Accounting (AAA) service based on Diameter protocol.
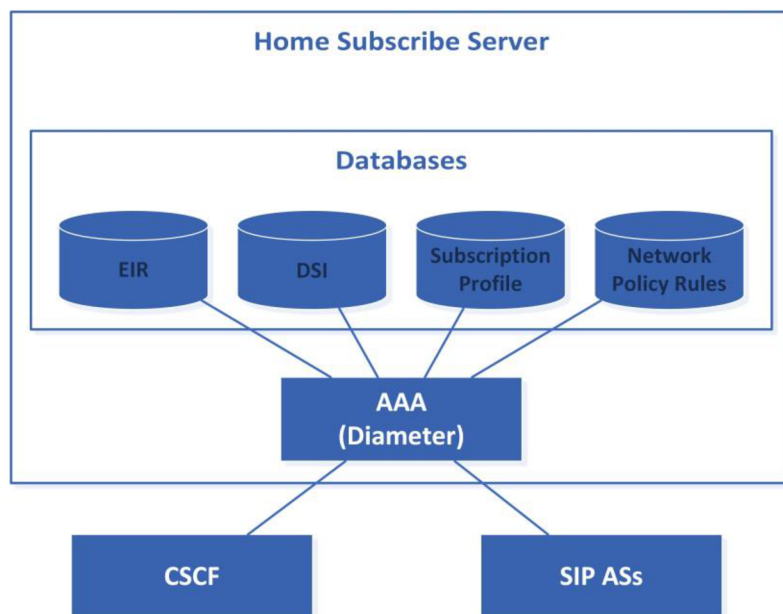
Picture 2-6 HSS structure

Packet Data Network Gateway (PGW) – default router for UE. Provide IP address allocation. Support policy and charging enforcement (PCEF), packet filtering for each user and performs marking for QoS management. UE may be connect to multiple PGW for accessing multiple Packet Data Networks. May coexist on same physical platform with SGW.

Serving gateway (SGW) routing and forwards packets between PGW and E-UTRAN, anchoring user plane during inter eNB handovers and as the anchor for mobility between LTE and other 3GPP technologies. Providing support for lawful interception. UE connected to one SGW.

The S5 interface provides user plane tunneling and tunnel management between SGW and PGW. The SGW and PGW may be realize as a single network element in which case the S5 interface is not exposed. (2)

Police and Charging Rules Function (PCRF) provides policy control decisions and flow based charging controls. Support for QoS in EPC network ensure that the user's plane traffic mapping and treatment is in accordance with the user's profile. Connected to PGW that implements policy and charging enforcement function (PCEF). The Gx interface is between the PCRF and the PGW, allowing the PCRF direct control over the policy enforcement functions of the PGW. The protocol used on the Gx interface is Diameter. (2)

Diameter Relay Agent DRA is a function specialized in forwarding Diameter messages, provide roaming to other LTE networks. Support for more MMEs and HSSs. (3) The DRA deployed between the EPC and IMS networks.

2.4 IP multimedia subsystem (IMS) architecture

IMS is the control infrastructure for supporting next generation IP Multimedia Services and consists of many separate elements.

Logically IMS can be divide by functional layers:

1. Control layer provides manage the session control, authorization, call routing and billing. The S-CSCF is the main call control element.
2. Service layer provides specific services (including telephony) implemented by Application Servers (AS). Once S-CSCF invokes a service, the AS can control the processing of the session
3. Media layer. Media processing – announcements servers, conference bridges, media servers – this layer is under control of the other two layers. Media Gateways are responsible for codec conversion. MGCF is physically implemented in 3G MSC

The **Call Session Control Function (CSCF)** may take on various roles as used in the IP multimedia subsystem: Proxy-CSCF (P-CSCF), Serving – CSCF and Interrogating-CSCF (I-CSCF)

Mw Interface (x-CSCF – x-CSCF)

The Mx interface is between a x-CSCF and another x-CSCF within the IMS core network (e.g. P-CSCF to I/S-CSCF). The protocols used on the Mw interface are SIP and SDP.
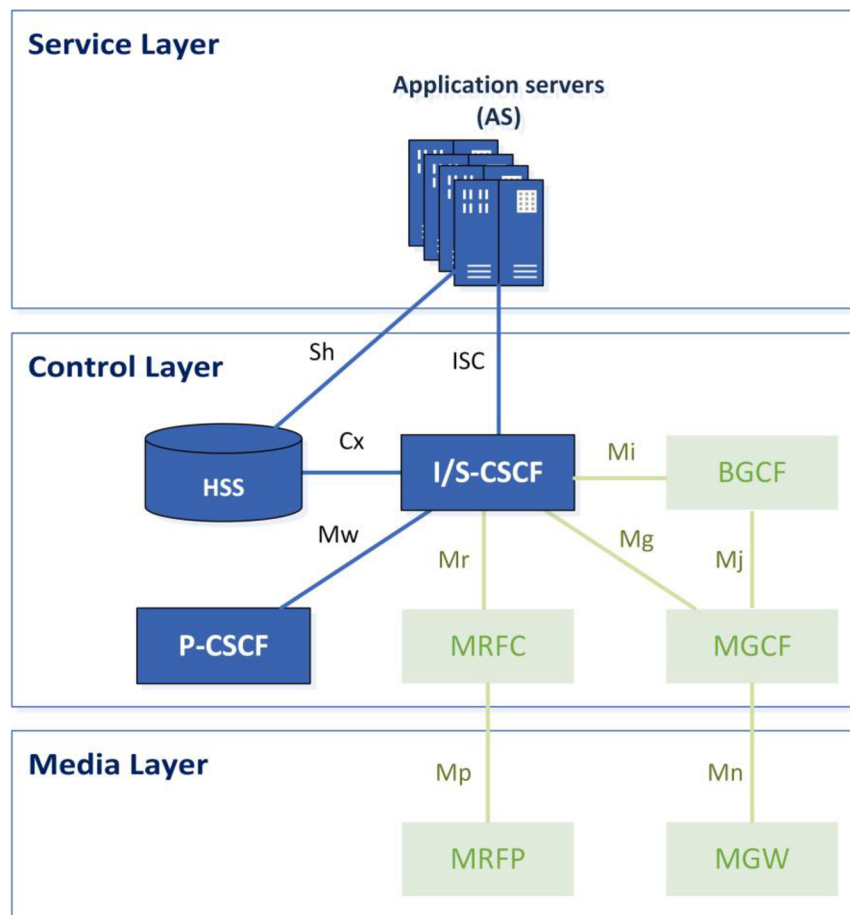
Picture 2-7 IMS layers

2.4.1 Proxy-CSCF (P-CSCF)

The **Proxy-CSCF** (P-CSCF) is the first contact point within the IM CN subsystem. Its address is discovered by UEs following PDP context activation, using the mechanism described in section "Procedures related to Local CSCF Discovery". The P-CSCF behaves like a Proxy (as defined in RFC2543 or subsequent versions), i.e. it accepts requests and services them internally or forwards them on, possibly after translation. The P-CSCF may also behave as a User Agent (as defined in the RFC2543 or subsequent versions), i.e. in abnormal conditions it may terminate and independently generate SIP transactions.

The Policy Control Function (PCF) is a logical entity of the P-CSCF. If the PCF was implement in a separate physical node, the interface between the PCF and the P-CSCF is not standardized.

The functions performed by the P-CSCF are:

1. Forward the SIP register request received from the UE to an I-CSCF determined using the home domain name, as provided by the UE.
2. Forward SIP messages received from the UE to the SIP server (e.g. S-CSCF) whose name the PCSCF has received as a result of the registration procedure.
3. As part of processing of the request and before forwarding, the P-CSCF may modify the Request URI of outgoing requests according to a set of provisioned rules defined

by the network operator (e.g. Number analysis and potential modification such as translation from local to international format.)

4. Forward the SIP request or response to the UE.
5. Detect an emergency session and select a S-CSCF in the visited network to handle emergency sessions. Interrogating – CSCF – is responsible for determining with which S-CSCF is the UE registered. The main functions of the I – CSCF are Select the S – CSCF during registration and load balancing across the S – CSCF.

Authorization of bearer resources and QoS management. Details of the P-CSCF role in QoS management and authorization of bearer resources for the session are being investigated by the QoS ad-hoc group. (6)

SGi Interface (PGW – P-CSCF)

The SGi interface is between the PGW and the P-CSCF within the IMS Network. The Gm reference point from the UE to P-CSCF is tunneled within SGi for VoLTE services. SGi is IP-based.

Rx Interface (PCRF – P-CSCF)

The Rx interface is between the appropriate Application Function (the P-CSCF in the case of VoLTE) and the PCRF allowing the Application Function to request the application of an appropriate policy for a session. The protocol used on the Rx interface is Diameter.

Gm Interface (UE – P-CSCF)

The Gm interface is between the UE and the P-CSCF and enables connectivity between the UE and the IMS network for registration, authentication, encryption, and session control. The protocol used on the Gm interface is SIP/SDP.

2.4.2 Interrogating-CSCF (I-CSCF)

Interrogating-CSCF (I-CSCF) is the contact point within an operator's network for all connections destined to a subscriber of that network operator, or a roaming subscriber currently located within that network operator's service area. There may be multiple ICSCFs within an operator's network.

The functions performed by the I-CSCF are:

1. Registration: Assigning a S-CSCF to a user performing SIP registration
2. Session Flows: Route a SIP request received from another network towards the S-CSCF, obtain from HSS the Address of the S-CSCF and forward the SIP request or response to the S-CSCF.

In performing the above functions, the operator may use the I-CSCF or other techniques to hide the configuration, capacity, and topology of the network from the outside. When the I-CSCF was chose to meet the hiding requirement then for sessions traversing across different operator's domains, the I-CSCF may forward the SIP request or response to another I-CSCF allowing the operators to maintain configuration independence. (6)

2.4.3 Serving – CSCF

Serving – CSCF – performs session management for subscriber's IMS based services. S-CSCF was select during SIP registration. Selection was basing on their ability to handle by authorized subscriber.

Within an operator's network, different S-CSCFs may have different functionalities. The functions performed by the S-CSCF during a session are:

Registration. May behave as a Registrar, accepts registration requests and makes its information available through the location server (e.g. HSS).

Session flows. Assure session control for the registered endpoint's sessions. Interact with Services Platforms for the support of services; provide endpoints with service event related information (e.g. notification of tones/announcement together with location of additional media resources, billing notification).

On behalf of an originating endpoint, obtain from a database the address of the I-CSCF for the network operator. When the destination subscriber is a customer of a different network operator, serving the destination subscriber from the destination name of the terminating subscriber (e.g. dial phone number or SIP URL), and forward the SIP request or response to that I-CSCF. When the destination name of the terminating subscriber (e.g. dial phone number or SIP URL), and the destination subscriber is a customer of the same network operator, forward the SIP request or response to an I-CSCF within the operator's network.

On behalf of a destination endpoint (i.e. the terminating subscriber/UE), forward the SIP request or response to a P-CSCF for a MT session to a home subscriber within the home network, or for a subscriber roaming within a visited network where the home network operator has chosen not to have an I-CSCF in the path. Forward the SIP request or response to an I-CSCF for a MT session for a roaming subscriber within a visited network where the home network operator has chosen to have an I-CSCF in the path. (6)

Cx Interface (I/S-CSCF – HSS)

The Cx interface is between the I/S CSCF and HSS to enable IMS registration and passing of subscriber data to the S-CSCF. The protocol used on the Cx interface is Diameter.

2.4.4 APN

Domain name system (DNS) functionality is used for mapping logical names to IP addresses in the mobile networks such a LTE in intra networks. It allows to configure aliases in IMS without linking to IP in IMS configuration files. Example of BIND9 based APN-DNS was in chapter 6.3.1.

2.4.5 TAS – Telephony Application Server

IMS support services subscribers using Application Services. Like another services VoLTE service need the Telephony AS. The critical moment in VoLTE implementation is that TAS must support licensed Adaptive Multirate Codec, or another codecs must convert by Multimedia Resurce Function.

Interfaces that TAS used to communicate with the other functions is Sh and ISC. The Sh interface is between the VoLTE Application Server and HSS to enable service and subscriber related information to be passed to the Application Server or stored in the HSS. The protocol used on the Sh interface is Diameter. The ISC interface is between S-CSCF and Telephony Application Server and is used to interact with the MMTel supplementary services implemented on the TAS. The protocol used on the ISC interface is SIP. The Cr Interface is between the Telephony Application Servers and the MRF. And is used for sending/receiving XML encoded media service requires (Cr) which are served by the MRF.

2.4.6 Additional multimedia support nodes

Multimedia Resource Function (MRF) is splitting into Multimedia Resource Function Controller (**MRFC**) and Multimedia Resource Function Processor (**MRFP**).

Tasks of the MRFC are the following:

Control the media stream resources in the MRFP.

Tasks of the MRFP include the following:

1. Control of the bearer on the Mb reference point.
2. Provide resources to be controlling by the MRFC.
3. Mixing of incoming media streams (e.g. for multiple parties).
4. Media stream source (for multimedia announcements).
5. Media stream processing (e.g. audio transcoding, media analysis).
6. Floor Control (i.e. manage access rights to shared resources in a conferencing environment).

Interpret information coming from an AS and S-CSCF (e.g. session identifier) and control MRFP accordingly. Application Server concerning MRF conference booking and manage booking information (e.g. start time, duration, and list of participants). (6)

3 Evolution and deployment

In chapter 3, I give a brief description of the evolution of mobile networks, LTE implementation phases, characteristics, and goals.

3.1 Evolution of mobile networks

Now 4G LTE is a newest mobile solution in the market. 3GPP evolution starts in 1999, when original UMTS was release. Then the other technologies was released that are using in 4G LTE:

Year	Release	Technology
1999	R99	Original UMTS
2000	R4	Basics of IMS
2001	R5	HSDPA, complete design of IMS
2002	R6	HSUPA, support for multimedia
2004	R7	HSPA+, MIMO
2007	R8	Defines the LTE and transition
2008	R9	Further enhancements of LTE
2011	R10	LTE Advanced
2013	R11	Continue of LTE Advanced – advanced interconnection between cervices

Table 3-1 3GPP evolution of technologies (1)

Historically, cellular technologies have adhered to an approximate 20-year cycle from launch to peak penetration, with around ten years between the launch of each new technology. The first commercial LTE networks went live in 2009 and based on historical precedent we would not expect the technology to reach a peak level of connections until around 2030.
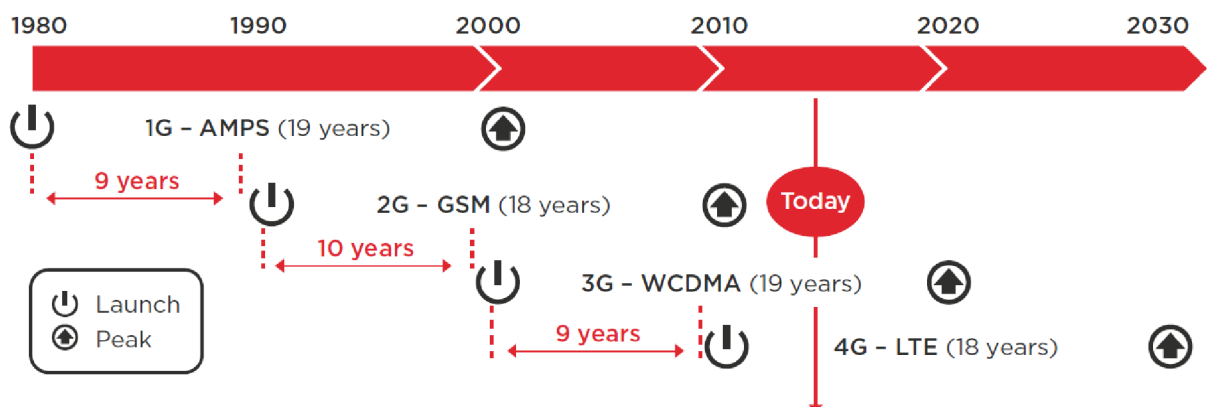
Figure 3-1 Evolution of mobile technology by generation, 1980 onwards (20)

These technologies are used to develop the final version of 4G LTE and deploying in three phases:

LTE Phase 1 – implementation EPC with oldies generation mobile networks. Uses for data services only.

LTE Phase 2 – voice implementation within 3G/2G – Circuit Switched Fall Back (CSFB).

LTE Phase 3 – Voice over LTE (VoLTE). Final phase, all mobile services handled by LTE network. Description of VoLTE released by GSM association (GSMA) in "VoLTE Service Description and Implementation Guidelines" in 2014. (2)

3.2 LTE characteristics and goals

Improved performance

- Efficient use of radio spectrum: UMTS had ~3 bps/Hz spectral utilization, LTE had 15 bps/Hz.
- High data rates: 100 Mbps downlink, 50 Mbps uplink
- Low latency: 10 ms and less, idle to activate transition under 100 ms
- Reduced costs
- Simplified architecture, minimum configuration. Only packet switched services was support, data services were shift away from 3G, leaving more capacity for Voice services in 3G.
- Less components delivering more services
- Increased Value
- Better user experience
- Allow access to services anytime, anywhere
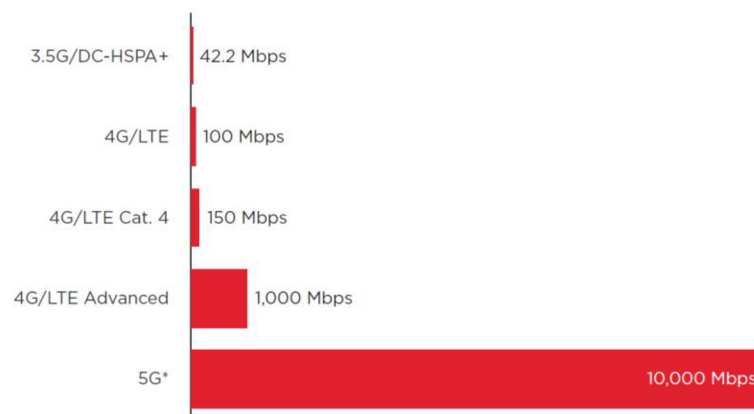
*Figure 3-2 Maximum theoretical downlink speed by technology generation, Mbps (*10 Gbps is the minimum theoretical upper limit speed specified for 5G) (20)*

4   Voice service

The modern networks are developed for providing higher data rates for subscribers. Implementation of new and expansion of the existing technologies are needed in cellular communication. The actual step in evolving is implementation of Voice over IP to 4G mobile networks.

In the specification 4G LTE by the 3rd Generation Partnership Project (3GPP) release 8, it was designed as a pure packet switched system and either data services and real time services would be carried by the IP protocol. However, 3GPP formally announced VoLTE in February 2010 and GSMA announced "VoLTE Service Description and Implementation Guidelines" in October 2014. During this time, the mobile communications service providers (CSPs) are necessary to use solutions to move a subscriber from LTE to a legacy technology to obtain circuit switched voice service – CSFB (circuit switched fallback). During fallback, the data rate is decreased because of using legacy data services too, for example GPRS, EDGE or HSDPA. In addition, in the latest generations handover increased call setup time twice.

In this chapter, I analyze the requirements of VoLTE implementation and specific implementation solutions.

4.1 VoLTE service requirements

Supporting voice over IP in a cellular communication system brings new challenges. Voice support in LTE requires the right mechanisms and architecture in radio and core networks, to guarantee quality of service and a good user experience because subscribers expect the same quality of service they know from circuit switched voice services, in GSM networks for example. Even after moving the voice services to VoIP, mobility between 4G and the previous cellular network generations is still the key problem.

4.1.1 The Adaptive Multi-Rate (AMR) speech codec

The AMR codec is originally developed and standardized by the European Telecommunications Standards Institute (ETSI) for GSM cellular systems. It is now chosen by the Third Generation Partnership Project (3GPP) as the mandatory codec for 3G and 4G cellular systems.

The AMR codec is a multi-mode codec that supports eight narrow band speech encoding modes with bit rates between 4.75 and 12.2 kbps. The sampling frequency used in AMR is 8000 Hz and the speech encoding is performed on 20 ms speech frames. Therefore, each encoded AMR speech frame represents 160 samples of the original speech.

Similar to AMR, the AMR-WB codec is also a multi-mode speech codec. AMR-WB supports nine wide band speech coding modes with respective bit rates ranging from 6.6 to 23.85 kbps. The sampling frequency used in AMR-WB is 16000 Hz and the speech processing is performed on 20 ms frames. This means that each AMR-WB encoded frame represents 320 speech samples. Both codecs support voice activity detection (VAD) and generation of comfort noise (CN) parameters during silence periods.

The main requirements of AMR

- UE has to support all modes
- If wideband AMR is available, it has to be supported too
- Both (UE and network) must support wideband modes 12.65, 8.85 and 6.6 kbps
- AMR coder takes 20 ms samples of speech and encodes them into frames
- Frames must be transferring across the network - RTP
- Delay between speech generation and reception should be below 150 ms else human can notice quality degradation
- License for using AMR

4.1.2 Robust Header Compression

UE and network must support Robust Header Compression (RoHC). The UE and network must be able to apply the compression to packets that were carry over the radio bearer dedicated for the voice media. At minimum, UE and network must support "RTP/UDP/IP" to compress RTP packets and "UDP/IP" to compress RTCP packets. The UE and network must support these profiles for both IPv4 and IPv6. (17)

4.1.3 LTE Radio Capabilities

LTE Signaling radio bearers (SRB) are used for the transfer of RRC and NAS signaling messages.

1. RRC messages are used as signaling between UE and eNodeB.
2. NAS (Non Access Stratum) messages are used as signaling between UE and MME. (18)

RRC messages can be used to encapsulate NAS messages for their transfer between UE and eNodeB. The S1 application protocol is later used to transfer NAS messages between eNodeB and MME. The UE must support the following combination of radio bearers for Voice over IMS:

SRB1 + SRB2 + 4 x AM DRB + 1 x UM DRB

The network must support the following combination of radio bearers:

SRB1 + SRB2 + 2 x AM DRB + 1 x UM DRB

One AM Data Radio Bearer (DRB) is utilized for Evolved Packet System (EPS) bearer with Quality of Service Class Indicator (QCI) = 5 and another AM DRB for EPS bearer with QCI = 8/9. UM DRB is utilized for EPS bearer with QCI = 1. (17)

LTE SRB Type	Direction	RRC message	RLC Mode
LTE SRB0 (CCCH)	Downlink	RRC Connection Setup RRC Connection Reject RRC Connection Re-establishment RRC Connection Re-establishment reject	Transparent
	Uplink	RRC Connection Request RRC Connection Re-embellishment Request	
LTE SRB1(DCCH)	Downlink	RRC Connection Reconfiguration RRC Connection Release Security Mode Command UE Capability Enquiry DL information transfer(if no SRB-2) Mobility from EUTRA Command Handover from EUTRA preparation request CS Fallback parameter response CDMA2000 Counter Check	
	Uplink	RRC Connection Setup Complete Security Mode Complete Security Mode Failure RRC Connection Reconfiguration Complete RRC Connection Re-establishment Complete Measurement report UE Capability information UL Information Transfer(if no SRB2) UL handover preparation transfer	

		CS fallback parameters request CDMA2000 Counter Check response	
LTE SRB2 (DCCH)	Downlink	DL Information Transfer	
	Uplink	UL Information Transfer	

Table 4-1 LTE Signaling Radio Bearer types (18)

4.1.4 RLC configurations

Radio Link Control (RLC) entity must be configured to perform data transfer in the following modes;

- Unacknowledged Mode (UM) for EPS bearers with QCI = 1
- Acknowledged Mode (AM) for EPS bearers with QCI = 5
- Acknowledged Mode (AM) for EPS bearers with QCI = 8/9

Voice service can tolerate error rates on the order of 1%, while benefiting from reduced delays, and is mapped to a radio bearer running the RLC protocol in unacknowledged mode (UM).

4.1.5 Guaranteed bit rate (GBR) bearer

Voice is one of the LTE services that require a guaranteed bit rate (GBR) bearer, although it is a very low data rate compared to LTE peak rates. The GBR bearer for voice requests dedicated network resources related to the Guaranteed Bit Rate (GBR) for AMR codec values. The network resources associated with the EPS bearer supporting GBR must be permanently allocated by admission control function in the eNodeB at bearer establishment. Reports from UE, including buffer status and measurements of UE's radio environment, must be required to enable the scheduling of the GBR. In UL it is the UE's responsibility to comply with GBR requirements. (17)
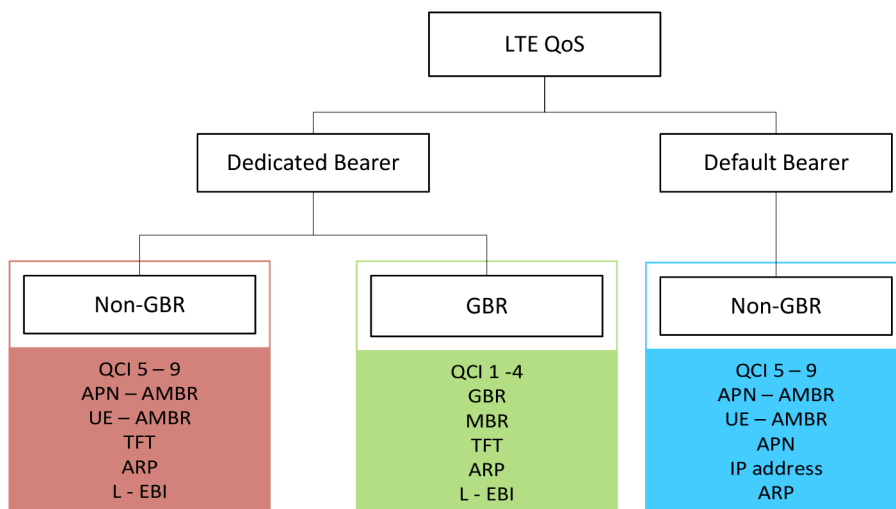
Figure 4-1 Bearer types and properties (19)

Dedicated bearer can be subdivided into Non-GBR and GBR types. GBR provides guaranteed bit rate and is associated with parameters like GBR and MBR

1. GBR: The minimum guaranteed bit rate per EPS bearer. Specified independently for uplink and downlink
2. MBR: The maximum guaranteed bit rate per EPS bearer. Specified independently for uplink and downlink

On the other hand, Non-GBR bearer does not provide guaranteed bit rate and has parameter like A- AMBR and UE- AMBR. (19)

1. A-AMBR: APN Aggregate maximum bit rate is the maximum allowed total non-GBR throughput to specific APN. It is specified interdependently for uplink an downlink
2. UE -AMBR: UE Aggregate maximum bit rate is the maximum allowed total non-GBR throughput among all APN to a specific UE

For an IMS session request for a Conversational Voice call (originating and terminating), a dedicated bearer for IMS-based voice must be created utilizing interaction with dynamic PCC. The network must initiate the creation of a dedicated bearer to transport the voice media. The dedicated bearer for Conversational Voice must utilize the standardized QCI value of one (1) and have the associated characteristics. (17)

The network must not create more than one dedicated bearer for voice media. Therefore, the UE and network must be able to multiplex the media streams from multiple concurrent voice sessions. (17)

For IMS session termination of a Conversational Voice call, the dedicated bearer must be deleted utilizing interaction with dynamic PCC. The network must initiate the deletion of the bearer.

4.1.6 P-CSCF Discovery

The UE and packet core must support the procedures for P-CSCF discovery via EPS. The UE shall indicate P-CSCF IPv6 Address Request and P-CSCF IPv4 Address Request when performing the following procedures:

1. During the initial attach when establishing PDN connection to the default APN,
2. During the initial attach when establishing PDN connection to the IMS well-known APN, and
3. During the establishment of the PDN connection to the IMS well-known APN when already attached,
4. During the attach procedure for emergency bearer services, and
5. During the establishment of the PDN connection for emergency bearer services when already attached.

The UE must use the P-CSCF addresses received during PDN connection establishment to the IMS well-known APN when accessing non-emergency services, and must use the P-CSCF addresses received during PDN connection establishment for emergency bearer services when accessing emergency services. (17)

4.1.7 Addressing

The UE and IMS core network must support Public User Identities, which includes all of the following types of addresses:

Alphanumeric SIP-URIs. Example: sip:voicemail@example.com

MSISDN represented as a SIP URI. Example: sip:+420774391777@example.com;user=phone

MSISDN represented as a Tel URI. Example: tel: :+420774391777

4.1.8 Handovers

- Handover are still needed for mobility
- eNB knows the QCI but not the service
- It can change process based on QCI
- Signaling is used to announce any changes
- "Data forwarding" feature – forwards data from old eNB to the new eNB while handover is in progress
- X2 and S1 handovers are not impacted by VoLTE

4.2 IMS protocols

For UE and IMS subsystem connectivity, in particular of voice services, important the following protocols:

- Session Initial Protocol (SIP)
- Session Description Protocol (SDP)
- Real-time Transport Protocol (RTP)
- RTP Control Protocol (RTCP)
- IP Security (IPsec)

4.2.1 Session Initial Protocol (SIP)

Session Initial Protocol is used to create, modify and terminate multimedia sessions. It is client-server signaling protocol, for delivering media tasks is used RTP/RTCP. SIP is a sequential request-response protocol. Every SIP request begins with a starting line that includes the name of the method (request type). A Request-Line contains a method name, a Request-URI, and the protocol version separated by a single space (SP) character. The Request-Line ends with CRLF.

Request-Line example: Method SP Request-URI SP SIP-Version CRLF

SIP Request Method	Description
INVITE	Indicates that a client is being invited to participate in a call session (7)
ACK	Confirms that the client has received a final response to an INVITE request (7)
BYE	Terminates a call; can be sent by either the caller or the called party (7)
CANCEL	Cancels any pending request (7)
OPTIONS	Queries the capabilities of servers (7)
REGISTER	Registers the address listed in the To header field with a SIP server (7)
PRACK	Provisional acknowledgement (8)
SUBSCRIBE	Subscribes to event notification (9)
NOTIFY	Notifies the subscriber of a new Event (9)
PUBLISH	Publishes an event to the Server (10)
INFO	Sends mid-session information that does not modify the session state (11)
REFER	Asks recipient to issue a SIP request (call transfer) (12)
MESSAGE	Transports instant messages using SIP (13)
UPDATE	Modifies the state of a session without changing the state of the dialog (14)

Picture 4-1 Request Methods description

SIP responses are distinguished from requests by having a Status-Line as their start-line. A Status-Line consists of the protocol version followed by a numeric Status-Code and its associated textual phrase, with each element separated by a single SP character.

Status-Line example: SIP-Version SP Status-Code SP Reason-Phrase CRLF

```
Request start line    INVITE sip:13@10.10.1.13 SIP/2.0
Request header        Via: SIP/2.0/UJP
                      10.10.1.99:5060;branch=z9hG4bK343b:628;rport
                      From: "Test 15" <sip:15@10.10.1.99>tag=as58f4201b
                      To: <sip:13@10.10.1.13>
                      Contact : <sip:15@10.10.1.99>
                      Call-ID: 326371826c80e17e6c:6c29861eb2933@10.10.1.99
                      CSeq: 102 INVITE
                      User-Agent : Asterisk PBX
                      Max-Forwards : 70
                      Date: Wed, 06 Dec 2009 14 :12 :45 GY.T
                      Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
                      SUBSCRIBE,
                      NOTIFY
                      Supported: replaces
                      Content-Type : application/adp
                      Content-Length: 258
<blank line>
Message body          v=0
(SDP message)         o=Joe Spirent 1821 1821 IN IP4 10.10.1.99
                      s=Spirent Seminar : IMS & VoLTE
                      c=IN IP4 10.10.1.99
                      t=0 0
                      m=audio 11424 RTP/AVP 0 8 101
                      a=rtpmap:0 PCMU/8000
                      a=rtpmap:8 PCMA/8000
                      a=rtpmap:101 telephone-event/8000
                      a=fmtp:101 0-16
                      a=silenceSupp:off - - - -
                      a=ptime:20
                      a=sendrecv
```

4.2.2 Session Description Protocol (SDP)

Session Description Protocol (SDP) – is used to describe multimedia sessions. SDP is contained in the body part of SIP. An SDP message is composed of a fields , whose names are abbreviated by a single lower-case letter. It conveys the name and purpose of the session, the media, protocols, codec formats, timing and transport information.

For example in the SDP message body describes the owner ("Joe Spirent"), the session ("Spirent Seminar: IMS &VoLTE"), some connection information (IP4 10.10.1.99), the media (audio) and some suggested attributes of the media (PCMU, PCMA, etc.). (15)

4.2.3 Real-time Transport Protocol (RTP) and RTP Control Protocol

IMS subsystem uses RTP as the media data transfer protocol. RTP provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. RTCP in IMS is used to provide Quality of Service (QoS) information and for synchronizing RTP streams. RTP and RTCP are always paired in port assignments. One port is used for audio data, and the other is used for control (RTCP) packets. For example: if RTP port equals "n", then RTCP port will be equal "n+1".

The audio conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration. Each chunk of audio data is preceded by an RTP header;

RTP header and data are in turn contained in a UDP packet. The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet so that senders can change the encoding during a conference, for example, to accommodate a new participant that is connected through a low-bandwidth link or react to indications of network congestion. (16)

The Internet, like other packet networks, occasionally loses and reorders packets and delays them by variable amounts of time. To cope with these impairments, the RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source, so that in this example, chunks of audio are contiguously played out the speaker every 20 ms. This timing reconstruction is performed separately for each source of RTP packets in the conference. The sequence number can also be used by the receiver to estimate how many packets are being lost. (16)

Since members of the working group join and leave during the conference, it is useful to know who is participating at any moment and how well they are receiving the audio data. For that purpose, each instance of the audio application in the conference periodically multicasts a reception report plus the name of its user on the RTCP (control) port. The reception report indicates how well the current speaker is being received and may be used to control adaptive encodings. In addition to the user name, other identifying information may also be included subject to control bandwidth limits. A site sends the RTCP BYE packet when it leaves the conference. (16)

5 VoLTE Implementation

In this chapter, I describe the main steps of VoLTE implementation at FEKT VUT.

5.1 EPC Configuration

5.1.1 Task Description

The VUT experimental LTE network was configured to support only data services, and it needs to activate VoLTE service only. In the next part I configure data in the scenario where VoLTE voice services are deployed in the EPS network, IMS interconnection is available between carriers, home subscribers and roaming subscribers subscribe to the IMS APN "IMS," and roaming subscribers use the local breakout mode to perform VoLTE voice services.
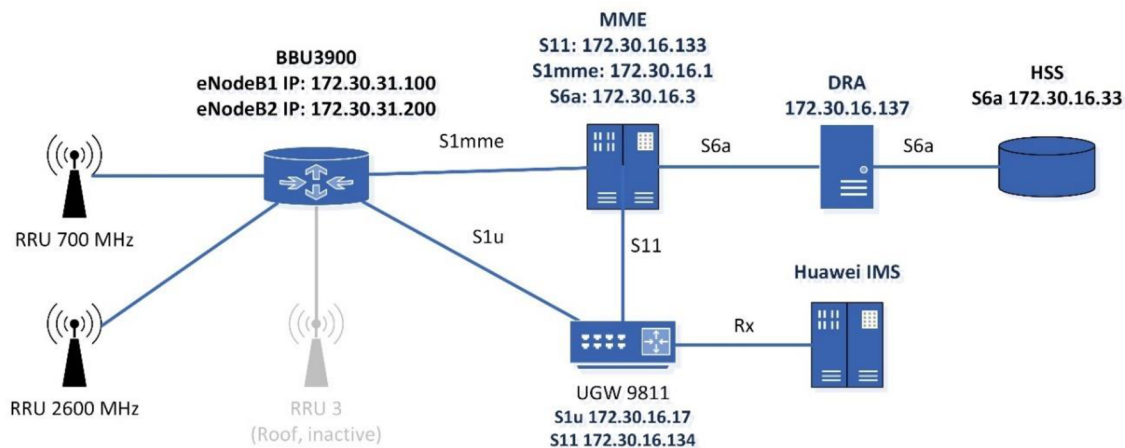
5.1.2 UGW Description and Configuration

The UGW9811 is a Huawei proprietary unified packet gateway. It is intended for use in the Enhanced 3G (E3G) Long Term Evolution/Evolved Packet Core (LTE/EPC) network. The UGW9811 can function as a GGSN, an S-GW, a P-GW, or any combination of them but can be conveniently managed as a single unified gateway.

Configure the context filter dedicated for IMS signaling with the destination address being the UE's IP address 172.30.20.6 , destination port number being 1 and 2, and protocol number being 6 and 17.

```
[UGW]apn ims
[UGW-apn-ims]ims-sig-filter 1 dest des-spe 172.30.20.6 dm-spe 255.255.255.255
dportop dp-spe 1 2 protop pro-spe 6

[UGW-apn-ims]ims-sig-filter 2 dest des-spe 172.30.20.6 dm-spe 255.255.255.255
dportop dp-spe 1 2 protop pro-spe 17
```

Then enable IMS function and save the configuration

```
[UGW-apn-ims]ims-switch enable
[UGW-apn-ims]reactivation-requested-delete enable
[UGW-apn-ims]temporarily-reject-retransmit enable
[UGW-apn-ims]quit
[UGW]softpara bit1310 1
[UGW]quit
<UGW>save
```

5.1.3 APN−based IMS services configuration

Enter the: system view, then APN view and configure the context filter dedicated for IMS signaling in a specified APN view.

```
system-view
apn
ims-sig-filter
```

In a specified APN view, enable the IMS, IMS signaling context, and IMS signaling air interface enhancement functions. During service processing, the configurations in the APN view are preferentially used, and the configurations in the PCC view are used only when inherit is configured in the APN view.

```
ims-switch
```

Configure whether the UGW9811 includes the reactivation flag in a default bearer deactivation message. When an abnormality occurs, the default bearer of a subscriber is deleted. If the subscriber does not access the network in time, the subscriber cannot receive service requests. After this command is executed, upon the deletion of a subscriber's default bearer, the UGW9811 initiates a reactivation procedure for the subscriber so that the subscriber accesses the network again.

Configure the caching and retransmission function for conflict scenarios.

```
temporarily-reject-retransmit
```

Return to the system view and then configure the default bearer/primary PDP context to create dedicated bearers for IMS signaling regardless of whether UEs contain an IMS signaling indication flag in the PCO IE.

```
reactivation-requested-delete
```

Currently, most UEs do not contain an IMS signaling indication flag in the PCO IE. If the software parameter BIT1310 is set to 0, the UGW9811 activates the UEs' context as ordinary contexts, which affects voice-related performance statistics.

```
softpara bit1310 1
```

Set the software parameter BIT909 to 1 if call holding is required.

```
softpara bit909 1
```

5.1.4 USN9810 MME configuration

Commands

Configure the APN data on the USN9810. In the MML Command - USN9810 window, run:

```
ADD IPV4DNSH: HSINDEX=Host Name Index, HOSTNAME=Host Name, ADDRSECTION=Address
Section, IPV4ADDR1=IP Address1, PRIORITY1=Priority1, WEIGHT1=Weight1;
ADD DNSN: FQDN=FQDN, HSINDEX=Host Name Index, ENTITY=Entity, INTYPE=Interface
Type, S5PROTOCOL=S5 Protocol, S8PROTOCOL=S8 Protocol, PRIORITY=Priority,
WEIGHT=Weight;
```

Then to configure the APN data on the DNS server are needed. In the workshop test lab we configured the DNS9816 device.

Then basic VoLTE services can be configured.

```
SET LICCTRL: PN="82204162", SWITCH=ON;
SET IMSVOPS: MMEHOMO=SUPPORT;
```

Activated bearer process configuration:

```
SET S1SMPROCTRL: PROT=DEDICATED_BEARER_ACT_PROC, CBRCONFLICTMODE=BUFFER;
SET S1SMPROCTRL: PROT=BEARER_MODIFICATION_PROC, UBRCONFLICTMODE=BUFFER;
```

Configure preferential selection of the shortest path for VoLTE services. MOD IMSIHLRHSS needs to be executed when the Gr interface does not support T-ADS query.

```
MOD IMSIHLRHSS: IMSIRANGE=IMSI Range, IMSIPRE=IMSI Prefix, INTRASINGLEREG=ENABLE,
INTRAFORCE=ULR, INTERSINGLEREG=ENABLE, INTERFORCE=ULR;
```

The next step is Voice Policy Control for User Group configuration. Turn on the switch for Voice Policy Control for User Group.

```
SET LICCTRL: PN="82205072", SWITCH=ON;
```

To enable all home subscribers to use the same voice service policy, perform the following operation:

```
ADD VOICEDEPLOY: SUBRANGE=HOME_USER, NOID=Network Operator Identity, IMSVOPS=YES,
APNNI=IMS APN Network Identifier, UETYPE=UE Type, CSFBNOTPREIND=CS Fallback not
preferred Indicator;
```

To enable home subscribers to use different voice service policies, for example, configuring VoLTE voice services for a small number of home subscribers at the initial deployment stage of VoLTE, perform the following operations:

```
ADD VOICEDEPLOY: SUBRANGE=HOME_USER, NOID=Network Operator Identity, IMSVOPS=NO;
ADD VOICEDEPLOY: SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, IMSVOPS=YES, APNNI=IMS
APN Network Identifier, UETYPE=UE Type, CSFBNOTPREIND=CS Fallback not preferred
Indicator;
```

Configure voice service policies for foreign subscribers. If the IMS APN subscribed by foreign subscribers and the IMS APN subscribed by home subscribers are different, for example, foreign subscribers use "IMS-X," and home subscribers use "IMS," and carriers do not require differentiation of CDRs by IMS APN, it is recommended that ADD ALIASAPN be executed to simplify network deployment.

According to GSMA specifications, VoLTE roaming services adopt the local breakout mode. If carriers require that the subscribers subscribing to the IMS APN "IMS" use the local breakout mode, ADD GWSELPLCY needs to be executed. If the local breakout mode is not forcedly used, this command is not executed, and the USN9810 determines whether to use the local breakout mode according to the description data of a foreign subscriber.

If alias APN conversion is not performed, APNNI in ADD GWSELPLCY must be set to the value of IMS APN Network Identifier added in ADD VOICEDEPLOY.

If alias APN conversion is performed, APNNI in ADD GWSELPLCY must be set to the value of APNNI Converted added in ADD ALIASAPN.

To enable all foreign subscribers to use the same voice service policy, perform the following operation:

```
ADD VOICEDEPLOY: SUBRANGE=HOME_USER, NOID=Network Operator Identity, IMSVOPS=NO;
ADD VOICEDEPLOY: SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, IMSVOPS=YES, APNNI=IMS
APN Network Identifier, UETYPE=UE Type, CSFBNOTPREIND=CS Fallback not preferred
Indicator;
```

Optional ADD ALIASAPN:

```
SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, OLDAPN=Original APNNI NEWAPN=APNNI
Converted;
Optional: ADD GWSELPLCY: SUBRANGE=FOREIGN_USER, NOID=Network Operator Identity,
APNNI=APNNI, VAA=YES;
```

To enable foreign subscribers to use different voice service policies, perform the following operations:

```
SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, OLDAPN=Original APNNI NEWAPN=APNNI
Converted;
Optional: ADD GWSELPLCY: SUBRANGE=FOREIGN_USER, NOID=Network Operator Identity,
APNNI=APNNI, VAA=YES;
```

ADD ALIASAPN:

```
ADD ALIASAPN: SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, OLDAPN=Original APNNI
NEWAPN=APNNI Converted;
```

ADD GWSELPLCY:

```
ADD GWSELPLCY: SUBRANGE=IMSI_PREFIX, IMSIPRE=IMSI Prefix, APNNI=APNNI, VAA=YES;
```

5.2 Tests on LTE RAN network

In this chapter was LTE RAN tests by Rohde & Schwarz R&S TMSW ROMES4 was image the FEKT VUT RAN bandwidths:

- 700 MHz (band 17)
- 1800 MHz (band 3)
- 2600 MHz (band 7)
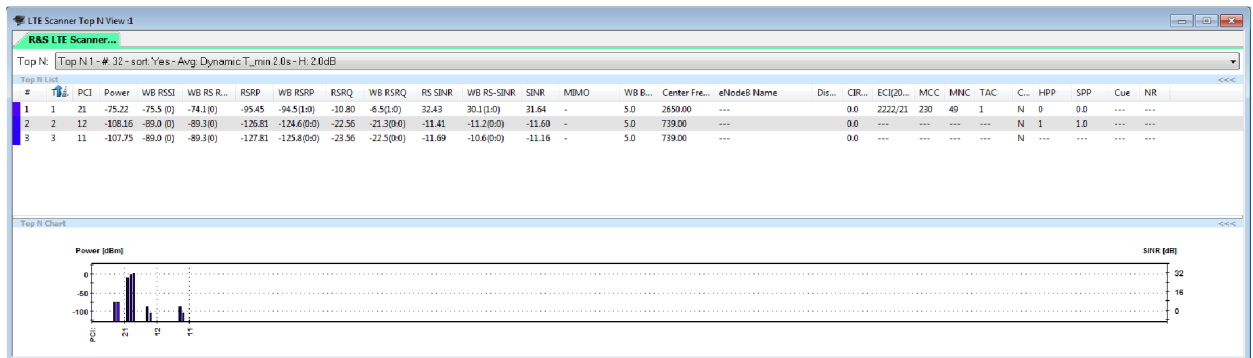
Picture 5-1 eNodeBs parameters

Band 17 - 700 MHz downlink

Band	Name	Downlink (MHz)			Bandwidth	Uplink (MHz)			Duplex spacing
		Low	Middle	High	DL/UL (MHz)	Low	Middle	High	(MHz)
		Earfcn				Earfcn			
17	700 b	734	**740**	746	12	704	**710**	716	30
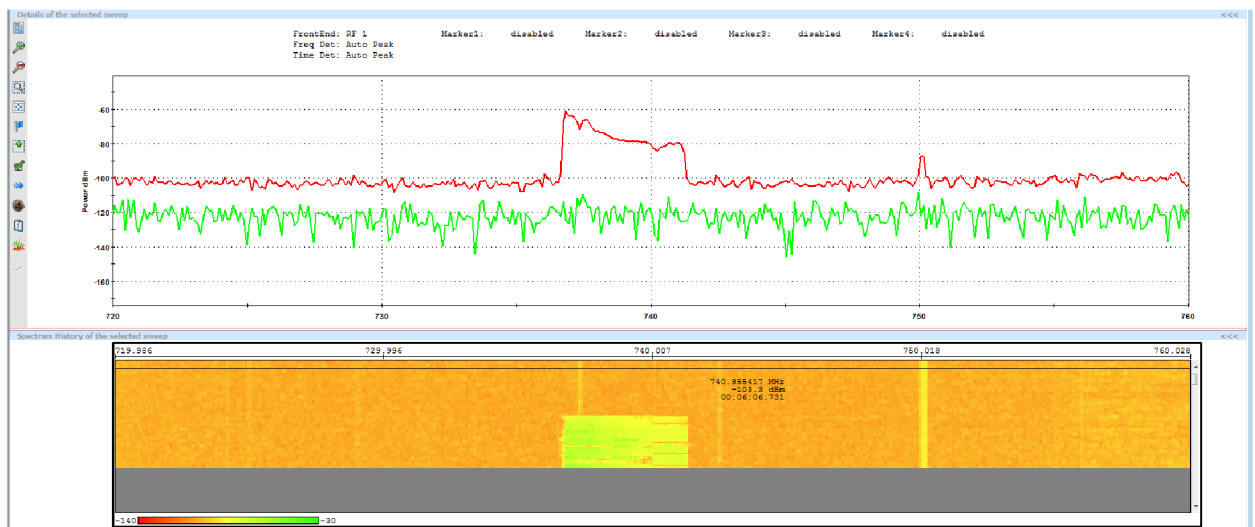
Picture 5-2 Band 17 - 700 MHz downlink

Band	Name	Downlink (MHz)			Bandwidth	Uplink (MHz)			Duplex spacing
		Low	Middle	High	DL/UL (MHz)	Low	Middle	High	(MHz)
		Earfcn				Earfcn			
7	2600	2620	2655	2690	70	2500	2535	2570	120
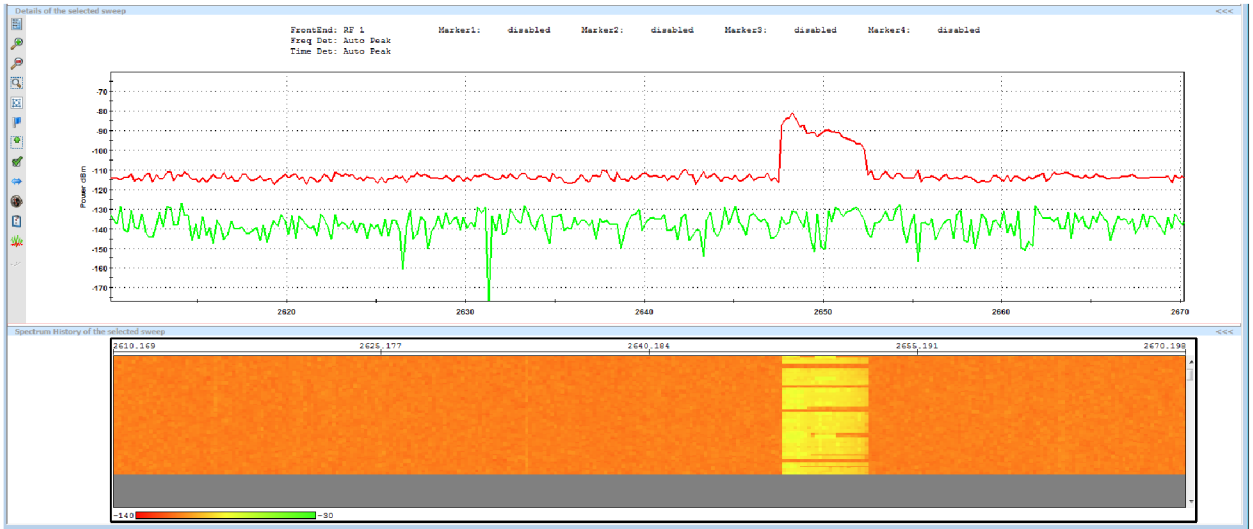
Picture 5-3 Band 7 - 2600 MHz downlink

5.3 EPC attach UE – eNodeB - MME

First step of the Attach UE to E-UTRAN is Random Access procedure. UE transfer the RRC Connection Request to eNodeB using SRB0 on the Common Control Channel (CCCH) for establish RRC connection. As the next step, the VoLTE UE initiates the Attach Request to the eNodeB with information including:

- EPS Attach Type
- NAS key set identifier
- IMSI
- UE network capability
- DRX parameters
- PDN Type (set to IPv4v6)
- PCO (P-CSCF IPv4 Address Request, P-CSCF IPv6 Address Request, IPv4 Link MTU Request)
- Voice Domain Preference and UE's Usage Setting (indicating support of IMS voice)
- ESM message container

The eNodeB selects the MME from the RRC parameters and forwards the Attach Request to the MME with the Selected Network and the TAI+ECGI location information of the cell where it received the message.

During the sniffing VoLTE network, UE's Attach Request resented between the eNodeB and MME was captured. Description and analyze of this packet was described in next part.

Communication is captured during the LTE UE attach. This request is received UE attach request between eNodeB and MME (S1AP protocol). Full protocol is in Appendix 1.

```
Frame 72: 190 bytes on wire (1520 bits), 190 bytes captured (1520 bits) on
interface0
Ethernet II, Src: HuaweiTe_7f:54:2e (7c:60:97:7f:54:2e), Dst: HuaweiTe_89:03:9b
(d8:49:0b:89:03:9b)
Internet Protocol Version 4, Src: 172.30.31.100, Dst: 172.30.16.1
```

The next parts of protocol will be changed in VoLTE.

1. EPS Attach Type:

```
.... .010 = EPS attach type: Combined EPS/IMSI attach (2)
```

2. NAS key set identifier

```
.001 .... = NAS key set identifier:  (1)
```

3. UE network capability

```
UE network capability
Length: 2
1... .... = EEA0: Supported
.1.. .... = 128-EEA1: Supported
..1. .... = 128-EEA2: Supported
...0 .... = 128-EEA3: Not Supported
.... 0... = EEA4: Not Supported
.... .0.. = EEA5: Not Supported
.... ..0. = EEA6: Not Supported
.... ...0 = EEA7: Not Supported
0... .... = EIA0: Not Supported
.1.. .... = 128-EIA1: Supported
..1. .... = 128-EIA2: Supported
...0 .... = 128-EIA3: Not Supported
.... 0... = EIA4: Not Supported
.... .0.. = EIA5: Not Supported
.... ..0. = EIA6: Not Supported
.... ...0 = EIA7: Not Supported
```

4. DRX parameters

```
DRX Parameter
                                    Element ID: 0x5c
                                    SPLIT PG CYCLE CODE: 10 (10)
                                    0000 .... = CN Specific DRX cycle
length coefficient: CN Specific DRX cycle length coefficient / value not specified
by the MS (0)
                                    .... 0... = SPLIT on CCCH: Split pg
cycle on CCCH is not supported by the mobile station
                                    .... .000 = Non-DRX timer: no non-DRX
mode after transfer state (0)
```

5. Voice Domain Preference and UE's Usage Setting (indicating support of IMS voice):

```
.... ..00 = Voice domain preference for E-UTRAN: CS Voice only (0)
```

6. ESM message container

```
ESM message container
Length: 33
ESM message container contents: 0204d011d1271a80802110010000108106000000000830600
```

7. TAI+ECGI location information of the cell where it received the message.

```
Item 2: id-TAI
      ProtocolIE-Field
      id: id-TAI (67)
      criticality: reject (0)
      value
            TAI
                  pLMNidentity: 32f094
                  Mobile Country Code (MCC): Czech Rep. (230)
                  Mobile Network Code (MNC): Unknown (49)
                  tAC: 0001 (1)
Item 3: id-EUTRAN-CGI
      ProtocolIE-Field
            id: id-EUTRAN-CGI (100)
            criticality: ignore (1)
            value
                  EUTRAN-CGI
```

Wireshark filter was used to clear capture:

```
not gtp and not tcp and not icmp and  ip.addr == 172.30.20.4 or ip.addr ==
172.30.20.5 or ip.addr == 172.30.20.6 or ip.addr == 172.30.31.100 or
ip.addr == 172.30.31.200 or ip.addr == 172.30.16.17 or ip.addr == 172.30.16.134 or
ip.addr ==
172.30.16.133 or ip.addr == 172.30.16.137 or ip.addr == 172.30.16.33 or ip.addr ==
172.30.16.34 or
ip.addr == 172.30.16.1 or ip.addr == 172.30.16.3
```

Full attach communication in Appendix2.

6 Laboratory exercise design

6.1 Overview

Theoretical and manual material are added as appendix to the diploma thesis. Here, I describe step by step the configurations and installation manuals.

For VoLTE support, I need to reconfigure EPC, IMS and update the UE's software (if supported). Because of other test and experiments with newly 4G network at FEKT VUT, I found another way.

1. Found the full supporting VoLTE service IMS.
2. Install the all needed functions in IMS, base configuration.
3. Configure both EPC and new IMS for working together.
4. Implement VoLTE service

This way help to avoid the changes in working IMS, it would looks like the EPC hire by the another provider. In addition, it would help students to working with the 4G network in future. Here is the key point to design the laboratory exercise.

In 3GPP documentation the Diameter routing agent is used for Diameter signal routing aggregation and applies to core network routing scenarios. GSMA define the DRA for security isolation in 3G/LTE inter-network roaming and interworking with legacy SS7 signaling networks.

It mean, that we can change actual VUT LTE network configuration to use Diameter agent as protocol between HSSs and other network components. It will allow to capture no ciphered signaling between mobile network elements.

Also exist opportunity to create for MKPM students fully accessed IMS. For it duplicate IMS with second HSS and DNS are needed.

Concept of these solutions will be presented at this part.

6.2 Software installation

For installation platform for virtual machines I choose the newest version of Ubuntu 16.04 (Xenial Xerus), Kernel Based Virtual Machine KVM and Virtual Machine Manager for GUI visualization. For the future signaling and resources separation was created four virtual machines and installed the next operation systems and software.
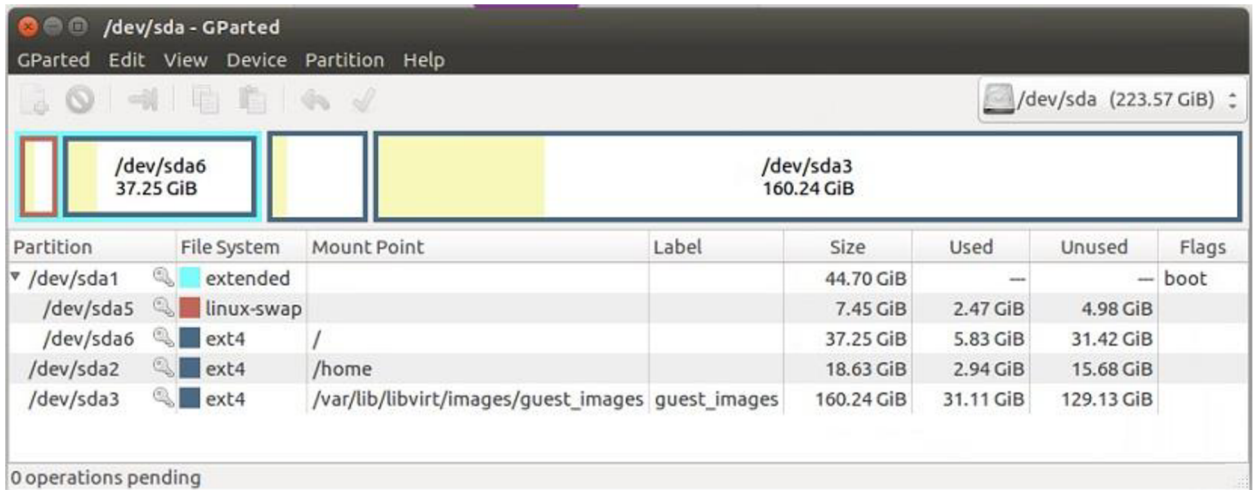
Picture 6-1 Partition for guest images
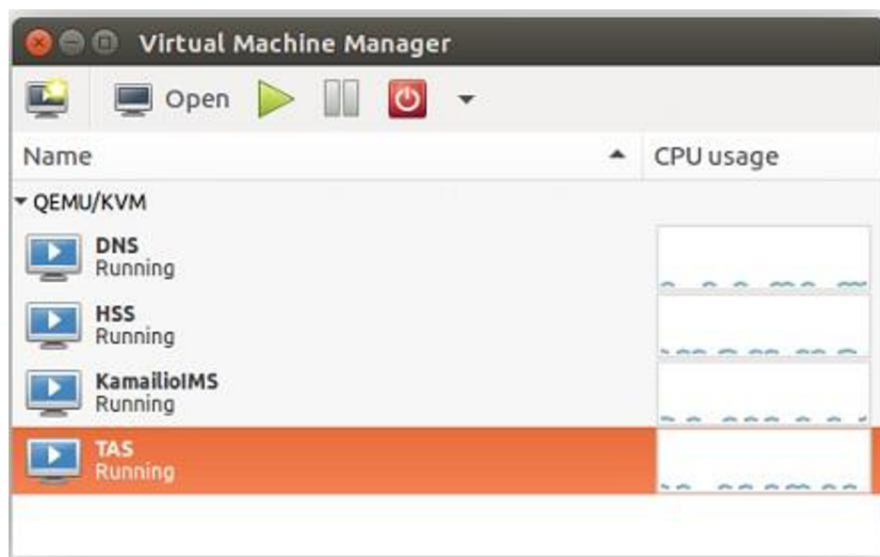
Picture 6-2 Virtual Machine Manager

6.2.1 Preparation

For the start, we need to add ng-voice.com key, repository and update all machines.

Add ng-voice.com key and repository and update:

```
wget -O - http://repository.ng-voice.com/PublicKey | apt-key add -
echo "deb http://repository.ng-voice.com wheezy ims rtpproxy fhoss sems" >
/etc/apt/sources.list.d/kamailio-ims.list
apt-get update
```

6.2.2 KamailioIMS VM installation

At this step I analyses the existing open source IMS projects, and choose the Kamailio IMS and FHOSS HSS. Kamailio is support VoLTE, Linux-based open source free IMS.

1) Kamailio IMS install:

```
apt-get install kamailio kamailio-ims-modules kamailio-presence-modules kamailio-
tls-modules kamailio-mysql-modules kamailio-xmlrpc-modules
```

2) NGCP-rtpengine - This is a module that enables media streams to be proxied via an RTP proxy.

```
apt-get install ngcp-rtpengine
```

6.2.3 HSS VM FHOSS installation

1) Next installing software, required for FHOSS: Oracle Java: for MySQL server and MySQL server: for database
2) FHOSS install. During installation process, dialog boxes would be opened.

```
apt-get install opnimscore-fhoss
```

3) FHOSS may be configured by web interface on port 8080. The default username/Password combination of hssAdmin/hss.

6.2.4 SEMS installation

SEMS - is a free, high performance, extensible media server for SIP based VoIP services. It would be the TAS (Telephony Application Server)

1) SEMS install:

```
apt-get install sems
```

6.2.5 DNS VM – BIND9 installation

BIND9 is available in the Main repository. No additional repository needs to be enabled for BIND9.

```
sudo apt-get install bind
```

6.2.6 Additional programs and modules

For the future work may be installed the next programs and modules:

- ngrep strives to provide most of GNU grep's common features, applying them to the network layer. ngrep is a pcap-aware tool that will allow you to specify extended regular or hexadecimal expressions to match against data payloads of packets.

```
apt-get install ngrep
```

- TCPDUMP

```
apt-get install tcpdump
```

- Libevent is an asynchronous event notification library that provides a mechanism to execute a callback function when a specific event occurs on a file descriptor or after a timeout has been reached.

```
apt-get install libevent-2.0-5
apt-get install libevent-pthreads-2.0-5
apt-get install libspandsp2
```

6.3 Configuration

The next table with network parameters, now are variables. All configuration files were analyzed and studied like it was needed in my task.

Parameters	Description
192.168.122.25	DNS dnslab IP NAT address
192.168.122.24	HSS
192.168.122.9	IMS
192.168.122.151	TAS
\<lte-apn.feec.vut.cz\>	must change to actual Huawei APN domain name.
\<lte-apn_IP\>	must change to actual Huawei APN IP address.
\<yyyymmddss\>	Serial Number: change date in format yyyymmdd, ss − number of change in common day.
\<feec_dns-IP\>	DNS server IP for internet access (192.168.110.41 or 192.168.110.38)
\<volte_lab_ims.feec.vutbr.cz\>	IMS alias for laboratory execution
\<volte_lab_ims-IP\>	IMS virtual machine IP for laboratory execution
\<volte-lab-AS_IP\>	AS virtual machine IP for laboratory execution
\<volte-lab-HSS_IP\>	HSS virtual machine IP for laboratory execution

6.3.1 BIND9 based APN configuration

Bind9 - Berkeley Internet Name Domain – one of the more popular, DNS. BIND9 Configuration files are stored in:

```
/etc/bind/
```

The main configuration is stored in the following files:

```
/etc/bind/named.conf
/etc/bind/named.conf.options
/etc/bind/named.conf.local
```

Address Records - the most commonly used type of record. This record maps an IP Address to a hostname. For example:

```
pcscf           5M IN A         <volte_lab_ims-IP>
```

Alias Records used to create an alias from an existing A record. You can create a CNAME record pointing to another CNAME record. But it doubles the number of requests made to the name server, thus making it an inefficient way to do so. For example:

```
www         6H IN A                 <feec_dns-IP>
<volte_lab_ims.feec.vutbr.cz>.        5M IN A         <volte_lab_ims-IP>
```

Name Server Records used to define which servers serve copies of this zone. This is where Primary and Secondary servers are defined.

```
        IN    NS    ns.example.com.
        [...]
ns      IN    A     1.2.3.4
```

```
$TTL 1H
@                       5M IN SOA               localhost.root.localhost. (
                                                <yyyymmddss>            ; serial
                                                5M                              ; refresh
                                                15M                             ; retry
                                                1W                              ; expiry
                                                5M )                            ; minimum
                    6H IN NSns1.<lte-apn.feec.vut.cz>.
ns1             6H IN A <lte-apn_IP>

; Websites / Public-Services:
www             6H IN A                 <feec_dns-IP>

<volte_lab_ims.feec.vutbr.cz>.         5M IN A            <volte_lab_ims-IP>

; <volte_lab_ims.feec.vutbr.cz>.       5M IN NAPTR 10 10 "s" "SIPS+D2T""""     _sips._tcp.pcscf
<volte_lab_ims.feec.vutbr.cz>.         5M IN NAPTR 10 30 "s" "SIP+D2U" ""      _sip._udp.pcscf
<volte_lab_ims.feec.vutbr.cz>.         5M IN NAPTR 10 20 "s" "SIP+D2T" ""      _sip._tcp.pcscf
```

```
pcscf                    5M IN A          <volte_lab_ims-IP>
pcscf                    5M IN NAPTR 10 10 "s" "SIP+D2T"        ""      _sip._tcp.pcscf
pcscf                    5M IN NAPTR 10 20 "s" "SIP+D2U"        ""      _sip._udp.pcscf
_sip._tcp.pcscf          5M SRV 10 1 4060 pcscf
_sip._udp.pcscf          5M SRV 10 1 4060 pcscf

icscf                    5M IN A          <volte_lab_ims-IP>
icscf                    5M IN NAPTR 10 50 "s" "SIP+D2U"        ""      _sip._udp.icscf
_sip._udp.icscf          5M SRV 20 0 5060 icscf

scscf                    5M IN A          <volte_lab_ims-IP>
scscf                    5M IN NAPTR 10 50 "s" "SIP+D2U"        ""      _sip._udp.scscf
_sip._udp.scscf          5M SRV 10 0 6060 scscf

as                       5M IN A          <volte-lab-AS_IP>
as                       5M IN NAPTR 10 50 "s" "SIP+D2U"        ""      _sip._udp.as
_sip._udp.as             5M SRV 10 0 7060 as

hss                      5M IN A          <volte-lab-HSS_IP>
```

6.3.2 Kamailio IMS configuration

Here is the simple modification pcscf.cfg, scscf.cfg that include IP-Addresses, Hostnames and Aliases.

```
### pcscf.cfg

# SIP / UDP
listen=udp:<volte_lab_ims-IP>:4060
listen=tcp:<volte_lab_ims-IP>:4060
# SIP / TCP
listen=tcp:127.0.0.1:4060
# SIP / TCP/TLS
#listen=tls:<volte_lab_ims-IP>:5061

alias=<pcscf.volte-lab-ims.feec.vut.cz>
alias=<proxy.volte-lab-ims.feec.vut.cz>
```

```
###scscf.cfg

# IP-Adress for incoming SIP-Traffic, in the following format:
listen=udp:46.101.144.112:6060
#listen=tcp:46.101.144.112:6060
listen=tcp:127.0.0.1:6060

#!define NETWORKNAME "kamailio-ims.org"
#!define NETWORKNAME_ESC "kamailio-ims\.org"
#!define HOSTNAME "scscf.kamailio-ims.org"
#!define URI "sip:scscf.kamailio-ims.org:6060"
#!define HOSTNAME_ESC "scscf\.kamailio-ims\.org"

alias=scscf.kamailio-ims.org
```

```
###scscf.cfg

# SIP / UDP
listen=udp:46.101.144.112:5060
# SIP / TCP
listen=tcp:127.0.0.1:5060
# SIP / TCP/TLS
#listen=tls:46.101.144.112:5061

alias=kamailio-ims.org
alias=icscf.kamailio-ims.org
```

The configuration for database for the Proxy, Serving and Interrogating CSCF was created by default, only need to configure the next:

After first configuration, executing the database for the Proxy-CSCF will be created.

```
CREATE DATABASE pcscf;
USE pcscf;
GRANT SELECT, INSERT, UPDATE, DELETE ON `pcscf`.* TO 'pcscf'@'localhost'
identified by 'heslo';
```

In kamailio.cfg was default working procedures. Must be define listen protocol and port, and change IP address to actual. Global parameters can be the same. In the last part, auto discovery of local aliases and DNS must be switched to on.

```
####### Defined Values #########
# *** Value defines - IDs used later in config

# SIP / UDP
listen=udp:46.101.144.112:6070

####### Global Parameters #########
#!ifdef WITH_DEBUG
debug=5
log_stderror=no
sip_warning=yes
children=4
#!else
debug=2
log_stderror=no
sip_warning=no
children=64
mem_safety=1
#!endif

user_agent_header="User-Agent: IMS-AS"
server_header="Server: IMS-AS"

/* comment the next line to enable the auto discovery of local aliases
   based on reverse DNS on IPs (default on) */
auto_aliases=no

disable_tcp=yes

check_via=no      # (cmd. line: -v)
dns=no            # (cmd. line: -r)
rev_dns=no        # (cmd. line: -R)
```

6.3.3 SIP Express Media Server (SEMS) configuration

SEMS - is a free, high performance, extensible media server for SIP based VoIP services. It would be the TAS (Telephony Application Server). The key moment why choose SEMS – AMR support. Patent licensing for AMR-NB is required.

SEMS was already prepared for working with IMS. We need to turn on in configuration and configure ipmap.conf.

Edit /etc/default/sems:

```
RUN_SEMS="yes"
```

Edit /etc/sems/etc/src_ipmap.conf

```
46.101.144.112=>mo
```

6.3.4 SIPWise' RTPEngine configuration

For RTP/RTCP functionality RTPEngine must be activated rtpengine-daemon:

Edit /etc/default/ngcp-rtpengine-daemon :

```
RUN_RTPENGINE=yes
[…]
INTERFACES="46.101.144.112"
[…]
```

7 Conclusion

In frame of this master thesis, I addressed four goals that was stated in the thesis assignment. The theoretical part of the work is given in the second, the third and the forth chapters. I gave the extended overview of the EPS-IMS system. I also described the mobile systems evolution from 1G to LTE and stated the main advantages of VoLTE technology. At last, I described the VoLTE solutions and formulated the VoLTE requirements.

In the practical part of the work I described the FEKT VUT VoLTE implementation. In addition, as a part of the practical part I created a Laboratory Exercise devoted to installation and configuration of IMS subsystem at FEKT VUT equipment.

The capture of UE registration to eNodeB was image the attach protocol flow, and protocol analyzer was representing the protocols parts, that will be changed in phase 3 implementation.

Additional, LTE RAN tests by Rohde & Schwarz R&S TMSW ROMES4 was image the RAN bandwidths of FEKT VUT is LTE 700 MHz (band 17) and 2600 MHz (band 7).

8 Literature

1. **3GPP.** Features and Study Items. *The 3rd Generation Partnership Project.* [Online] http://www.3gpp.org/DynaReport/FeatureListFrameSet.htm.

2. **GSM Association.** VoLTE Service Description and Implementation Guidelines Version 2.0. *http://www.gsma.com/.* [Online] October 7, 2014. http://www.gsma.com/network2020/wp-content/uploads/2014/10/FCM.01-VoLTE-Service-Description-and-Implementation-Guidelines-Version-2.0.pdf.

3. **Association, GSM.** LTE Roaming Guidelines Version 9.0. *www.gsma.com.* [Online] January 24, 2013. http://www.gsma.com/newsroom/wp-content/uploads/2013/04/IR.88-v9.0.pdf.

4. **Frédéric Firmin, 3GPP MCC.** NAS. Overview of the Non-Access Stratum. *http://www.3gpp.org/.* [Online] http://www.3gpp.org/technologies/keywords-acronyms/96-nas.

5. **3GPP.** S1 Application Protocol (S1AP) 3GPP TS 36.413 version 12.3.0 Release 12. *www.etsi.org.* [Online] http://www.etsi.org/deliver/etsi_ts/136400_136499/136413/12.03.00_60/ts_136413v120300p.pdf.

6. —. IP Multimedia Subsystem (IMS) Stage 2 3GPP TS 23.228 version 13.5.0 Release 13. *www.3gpp.org.* [Online] april 2016. http://www.etsi.org/deliver/etsi_ts/123200_123299/123228/13.05.00_60/ts_123228v130500p.pdf.

7. **J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler.** RFC3261 SIP: Session Initiation Protocol. *tools.ietf.org.* [Online] June 2002. https://tools.ietf.org/html/rfc3261.

8. **J. Rosenberg, H. Schulzrinne.** RFC3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP). *tools.ietf.org.* [Online] June 2002. https://tools.ietf.org/html/rfc3262#section-5.

9. **A. B. Roach.** RFC3265 Session Initiation Protocol (SIP)-Specific Event Notification. *tools.ietf.org.* [Online] June 2002. https://tools.ietf.org/html/rfc3265.

10. **A. Niemi.** RFC3903 Session Initiation Protocol (SIP) Extension for Event State Publication. *tools.ietf.org.* [Online] October 2004. https://tools.ietf.org/html/rfc3903.

11. **C. Holmberg, E. Burger, H. Kaplan.** RFC6086 Session Initiation Protocol (SIP) INFO Method and Package Framework. *tools.ietf.org.* [Online] January 2011. https://tools.ietf.org/html/rfc6086.

12. **R. Sparks.** RFC3515 The Session Initiation Protocol (SIP) Refer Method. *tools.ietf.org.* [Online] April 2003. https://tools.ietf.org/html/rfc3515.

13. **B. Campbell, J. Rosenberg, H. Schulzrinne, C. Huitema, D. Gurle.** RFC3428 Session Initiation Protocol (SIP) Extension for Instant Messaging. *www.ietf.org.* [Online] December 2002. https://www.ietf.org/rfc/rfc3428.txt.

14. **J. Rosenberg.** RFC3311 The Session Initiation Protocol (SIP) UPDATE Method. *tools.ietf.org.* [Online] September 2002. https://tools.ietf.org/html/rfc3311.

15. **SPIRENT.** IMS Procedures and Protocols. *http://www.spirent.com/.* [Online] 2014. http://www.spirent.com/~/media/White%20Papers/Mobile/Protocol_Reference_Guide.pdf.

16. **H. Schulzrinne, S. Casner, R. Frederick, V. Jacobson.** RFC3550 RTP: A Transport Protocol for Real-Time Applications. *tools.ietf.org.* [Online] July 2003. https://tools.ietf.org/pdf/rfc3550.pdf.

17. **Association, GSM.** IMS Profile for Voice and SMS IR.92 Version 9.0. *http://www.gsma.com/.* [Online] April 8, 2015. http://www.gsma.com/newsroom/wp-content/uploads/IR.92-v9.0.pdf.

18. **RF Wireless World.** LTE Signalling Radio Bearer types-SRB0,SRB1,SRB2. *http://www.rfwireless-world.com/.* [Online] http://www.rfwireless-world.com/Tutorials/LTE-signalling-radio-bearers.html.

19. **4g-lte-world.** Quality of Service (QoS) in LTE. *http://4g-lte-world.blogspot.cz/.* [Online] January 2013. http://4g-lte-world.blogspot.cz/2013/01/quality-of-service-qos-in-lte.html.

20. **GSM Association: Dan Warren, Calum Dewar.** Understanding 5G: Perspectives on future technological advancements in mobile. *https://www.gsmaintelligence.com.* [Online] December 2014. https://www.gsmaintelligence.com/research/?file=141208-5g.pdf&download.

21. **3GPP.** IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); 3GPP TS 24.229 version 13.5.1 Release 13. *http://www.3gpp.org/.* [Online] 05 2016. http://www.etsi.org/deliver/etsi_ts/124200_124299/124229/13.05.01_60/ts_124229v130501p.pdf.

9 Abbreviations

3GPP	3rd Generation Partnership Project
A-SBC	Access Session Border Controller
ACR	Anonymous Call Rejection
AMBR	Aggregate Maximum Bit Rate
AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
APN	Access Point Name
ARP	Allocation and Retention Priority
AS	Application Server
ATCF	Access Transfer Control Function
ATGW	Access Transfer Gateway
ATU-STI	Access Transfer Update – Session Transfer ID
AUTN	Authentication Token
AVP	Attribute Value Pair
BGCF	Border Gateway Control Function
BICC	Bearer Independent Call Control
CDIV	Communication Diversion
CDR	Charging Data Record
CN	Core Network
CONF	Conferencing
CS	Circuit Switched
CSCF	Call Server Control Function
CSFB	Circuit Switched Fall Back
DEA	Diameter Edge Agent
DL	DownLink
DNS	Domain Name System
DPI	Deep Packet Inspection
DRA	Diameter Relay Agent
DRX	Discontinuous Reception
DSCP	DiffServ Code Point
ECGI	E-UTRAN Cell Global Identifier
eKSI	E-UTRAN Key Set Identifier
ENUM	E.164 Number Mapping
EPC	Evolved Packet Core
EPS	Evolved Packet System
ERAB	E-UTRAN Radio Access Bearer
ESM	EPS Session Management
ETSI	European Telecommunications Standards Institute
E-UTRAN	Evolved Universal Terrestrial Access Network
FDD	Frequency Division Duplex
GAA	Generic Authentication Architecture
GBA	Generic Bootstrapping Architecture
GBR	Guaranteed Bit Rate
GPRS	General Packet Radio Service

GRX	GPRS Roaming eXchange
GSM	Global System for Mobile communications
GTP	GPRS Tunnelling Protocol
HLR	Home Location Register
HPMN	Home Public Mobile Network
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
HTTP	Hypertext Transfer Protocol
I-CSCF	Interrogating Call Session Control Function
I-SBC	Interconnect Session Border Controller
IBCF	Interconnection Border Control Function
ICID	IM CN subsystem charging identifier
ICS	IMS Centralised Services
ICSI	IMS Communication Service Identifier
IM	IP Multimedia
IM-GW	IP Media Gateway
IMEI	International Mobile Equipment Identity
IMS	IP Multimedia Subsystem
IMS-AKA	IMS Authentication and Key Agreement
IMS-AGW	IMS Access Gateway
IMS-ALG	IMS Application Level Gateway
IMSI	International Mobile Subscriber Identity
IOT	Interoperability Testing
IP	Internet Protocol
IP-CAN	IP-Connectivity Access Network
IPsec	IP Security
IPX	IP Packet Exchange
ISIM	IM Services Identity Module
ISUP	ISDN User Part
LBO	Local Breakout
LTE	Long Term Evolution
MAC	Medium Access Control
MBR	Maximum Bit Rate
MCC	Mobile Country Code
ME	Mobile Equipment
MGCF	Media Gateway Control Function
MME	Mobility Management Entity
MMS	Multimedia Messaging Service
MMTel	Multimedia Telephony
MNC	Mobile Network Code
MNO	Mobile Network Operator
MRF	Media Resource Function
MSC	Mobile Switching Centre
MSISDN	Mobile Subscriber ISDN Number
MSRP	Message Session Relay Protocol
MTU	Maximum Transmission Unit
MWI	Message Waiting Indicator
NAPTR	Name Authority Pointer
NAS	Non-Access Stratum

NAT	Network Address Translation
NNI	Network to Network Interface
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
P-CSCF	Proxy Call Session Control Function
PCC	Policy and Charging Control
PCEF	Policy and Charging Enforcement Function
PCO	Protocol Configuration Options
PCRF	Policy Charging and Rules Function
PDN	Packet Data Network
PGW	Packet Data Network Gateway
PHB	Per Hob Behaviour
PLMN	Public Land Mobile Network
PMN	Public Mobile Network
PS	Packet Switched
QCI	QoS Class Identifier
QoS	Quality of Service
RAB	Radio Access Bearer
RAN	Radio Access Network
RAND	RANDom number (used for authentication)
RAT	Radio Access Technology
RES	user RESponse (used in IMS-AKA)
RLC	Radio Link Control
RRC	Radio Resource Control
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
S-CSCF	Serving Call Session Control Function
SAE	System Architecture Evolution
SBC	Session Border Controller
SCC AS	Service Centralization & Continuity AS
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SEG	Security Gateway
SGW	Serving Gateway
SGSN	Serving GPRS Support Node
SIGCOMP	Signalling Compression
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SMS	Short Message Service
SON	Self-Organising Networks
SRTP	Secure RTP
SRVCC	Single Radio Voice Call Continuity
STN-SR	Session Transfer Number for SRVCC
TAS	Telephony Application Server
TAI	Tracking Area Identity
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplexing
TEID	Tunnel End Point Identifier

TFT	Traffic Flow Template
THP	Traffic Handling Priority
TLS	Transport Layer Security
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TrGW	Transition Gateway
TTM	Time To Market
UDC	User Data Convergence
UDP	User Datagram Protocol
UDR	User Data Repository
UE	User Equipment
UICC	Universal Integrated Circuit Card
UL	Uplink
ULI	User Location Information
UMTS	Universal Mobile Telecommunications System
UNI	User to Network Interface
URN	Uniform Resource Name
URI	Uniform Resource Identifier
USIM	Universal Subscriber Identity Module
UTRAN	Universal Terrestrial Access Network
VLR	Visitor Location Register
VoLTE	Voice over LTE
XRES	eXpected user RESponse (used in IMS-AKA)

Appendices

Appendix 1. E-NodeB attach request protocols

```
Frame 72: 190 bytes on wire (1520 bits), 190 bytes captured (1520 bits) on
interface 0
Ethernet II, Src: HuaweiTe_7f:54:2e (7c:60:97:7f:54:2e), Dst:
HuaweiTe_89:03:9b (d8:49:0b:89:03:9b)
Internet Protocol Version 4, Src: 172.30.31.100, Dst: 172.30.16.1
    0100 .... = Version: 4
    .... 0101 = Header Length: 20 bytes
    Differentiated Services Field: 0xb8 (DSCP: EF PHB, ECN: Not-ECT)
    Total Length: 176
    Identification: 0x5a55 (23125)
    Flags: 0x00
    Fragment offset: 0
    Time to live: 254
    Protocol: SCTP (132)
    Header checksum: 0xd91a [validation disabled]
    Source: 172.30.31.100
    Destination: 172.30.16.1
    [Source GeoIP: Unknown]
    [Destination GeoIP: Unknown]
Stream Control Transmission Protocol, Src Port: 36412 (36412), Dst Port:
36412 (36412)
S1 Application Protocol
    S1AP-PDU: initiatingMessage (0)
        initiatingMessage
            procedureCode: id-initialUEMessage (12)
            criticality: ignore (1)
            value
                InitialUEMessage
                    protocolIEs: 5 items
                        Item 0: id-eNB-UE-S1AP-ID
                            ProtocolIE-Field
                                id: id-eNB-UE-S1AP-ID (8)
                                criticality: reject (0)
                                value
                                    ENB-UE-S1AP-ID: 168444
                        Item 1: id-NAS-PDU
                            ProtocolIE-Field
                                id: id-NAS-PDU (26)
                                criticality: reject (0)
                                value
                                    NAS-PDU:
17c5e4d695830741120bf632f094800101c702001c02e060...
                                    Non-Access-Stratum (NAS)PDU
                                        0001 .... = Security header type:
Integrity protected (1)
                                        .... 0111 = Protocol discriminator:
EPS mobility management messages (0x07)
                                        Message authentication code:
0xc5e4d695
                                        Sequence number: 131
                                        0000 .... = Security header type:
Plain NAS message, not security protected (0)
```

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```
                                        .... 0111 = Protocol discriminator:
EPS mobility management messages (0x07)
                                        NAS EPS Mobility Management Message
Type: Attach request (0x41)
                                        0... .... = Type of security context
flag (TSC): Native security context (for KSIasme)
                                        .001 .... = NAS key set identifier:
(1)
                                        .... 0... = Spare bit(s): 0x00
                                        .... .010 = EPS attach type: Combined
EPS/IMSI attach (2)
                                        EPS mobile identity
                                            Length: 11
                                            .... 0... = odd/even indic: 0
                                            .... .110 = Type of identity:
GUTI (6)
                                            Mobile Country Code (MCC): Czech
Rep. (230)
                                            Mobile Network Code (MNC):
Unknown (49)
                                            MME Group ID: 32769
                                            MME Code: 1
                                            M-TMSI: 0xc702001c
                                        UE network capability
                                            Length: 2
                                            1... .... = EEA0: Supported
                                            .1.. .... = 128-EEA1: Supported
                                            ..1. .... = 128-EEA2: Supported
                                            ...0 .... = 128-EEA3: Not
Supported
                                            .... 0... = EEA4: Not Supported
                                            .... .0.. = EEA5: Not Supported
                                            .... ..0. = EEA6: Not Supported
                                            .... ...0 = EEA7: Not Supported
                                            0... .... = EIA0: Not Supported
                                            .1.. .... = 128-EIA1: Supported
                                            ..1. .... = 128-EIA2: Supported
                                            ...0 .... = 128-EIA3: Not
Supported
                                            .... 0... = EIA4: Not Supported
                                            .... .0.. = EIA5: Not Supported
                                            .... ..0. = EIA6: Not Supported
                                            .... ...0 = EIA7: Not Supported
                                        ESM message container
                                            Length: 33
                                            ESM message container contents:
0204d011d1271a8080211001000010810600000000830600...
                                        Tracking area identity - Last visited
registered TAI
                                            Element ID: 0x52
                                            Mobile Country Code (MCC): Czech
Rep. (230)
                                            Mobile Network Code (MNC):
Unknown (49)
                                            Tracking area code(TAC): 0x0001
                                        DRX Parameter
                                            Element ID: 0x5c
                                            SPLIT PG CYCLE CODE: 10 (10)
```

0000 .... = CN Specific DRX cycle
length coefficient: CN Specific DRX cycle length coefficient / value not
specified by the MS (0)
                                    .... 0... = SPLIT on CCCH: Split
pg cycle on CCCH is not supported by the mobile station
                                    .... .000 = Non-DRX timer: no
non-DRX mode after transfer state (0)
                            TMSI Status
                                    1001 .... = Element ID: 0x9-
                                    .... 000. = Spare bit(s): 0
                                    ....0 = TMSI flag: no valid
TMSI available
                            Mobile station classmark 2
                                    Element ID: 0x11
                                    Length: 3
                                    0... .... = Spare: 0
                                    .10. .... = Revision Level: Used
by mobile stations supporting R99 or later versions of the protocol (2)
                                    ...0 .... = ES IND: Controlled
Early Classmark Sending option is not implemented in the MS
                                    .... 1... = A5/1 algorithm
supported: encryption algorithm A5/1 not available
                                    .... .111 = RF Power Capability:
RF Power capability is irrelevant in this information element (7)
                                    0... .... = Spare: 0
                                    .0.. .... = PS capability
(pseudo-synchronization capability): PS capability not present
                                    ..01 .... = SS Screening
Indicator: Capability of handling of ellipsis notation and phase 2 error
handling (1)
                                    .... 1... = SM capability (MT SMS
pt to pt capability): Mobile station supports mobile terminated point to
point SMS
                                    .... .0.. = VBS notification
reception: no VBS capability or no notifications wanted
                                    .... ..0. = VGCS notification
reception: no VGCS capability or no notifications wanted
                                    ....0 = FC Frequency
Capability: The MS does not support the E-GSM or R-GSM band
                                    1... .... = CM3: The MS supports
options that are indicated in classmark 3 IE
                                    .0.. .... = Spare: 0
                                    ..1. .... = LCS VA capability
(LCS value added location request notification capability): LCS value added
location request notification capability supported
                                    ...0 .... = UCS2 treatment: the
ME has a preference for the default alphabet
                                    .... 0... = SoLSA: The ME does
not support SoLSA
                                    .... .1.. = CMSP: CM Service
Prompt: Network initiated MO CM connection request supported for at least one
CM protocol
                                    .... ..1. = A5/3 algorithm
supported: encryption algorithm A5/3 available
                                    ....0 = A5/2 algorithm
supported: encryption algorithm A5/2 not available
                            Additional update type
                                    1111 .... = Element ID: 0xf-
                                    .... 000. = Spare bit(s): 0x00

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```
                                                       .... ...1 = AUTV: SMS only (1)
                                             Voice domain preference and UE's
usage setting

                                             Element ID: 0x5d
                                             Length: 1
                                             0000 0... = Spare bit(s): 0
                                             .... .0.. = UE's usage setting:
Voice centric

                                             .... ..00 = Voice domain
preference for E-UTRAN: CS Voice only (0)
                                         GUTI type - Old GUTI type
                                             1110 .... = Element ID: 0xe-
                                             .... 000. = Spare bit(s): 0x00
                                             .... ...0 = GUTI type: Native
GUTI
                            Item 2: id-TAI
                                ProtocolIE-Field
                                    id: id-TAI (67)
                                    criticality: reject (0)
                                    value
                                        TAI
                                            pLMNidentity: 32f094
                                            Mobile Country Code (MCC): Czech Rep.
(230)
                                            Mobile Network Code (MNC): Unknown
(49)
                                            tAC: 0001 (1)
                            Item 3: id-EUTRAN-CGI
                                ProtocolIE-Field
                                    id: id-EUTRAN-CGI (100)
                                    criticality: ignore (1)
                                    value
                                        EUTRAN-CGI
                            Item 4: id-RRC-Establishment-Cause
                                ProtocolIE-Field
                                    id: id-RRC-Establishment-Cause (134)
                                    criticality: ignore (1)
                                    value
                                        RRC-Establishment-Cause: mo-Signalling
(3)
```
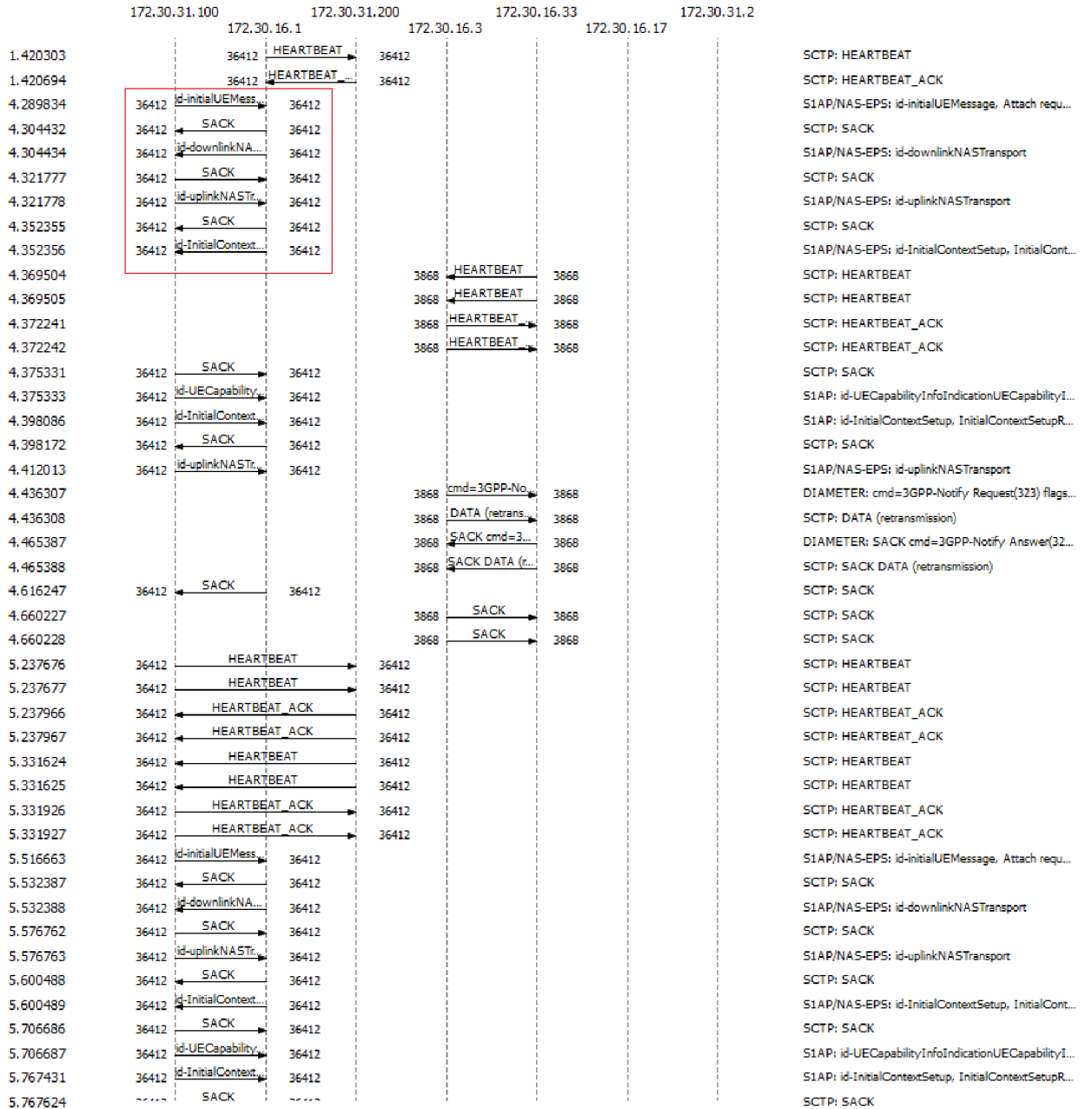
Appendix 2. UE to E-NodeB attach request (E-NodeB – MME communication)

S1MME IP address: 172.30.16.1

eNodeB IP address 172.30.31.100

Appendix 3. VoLTE architecture