VOICE OVER LTE (VOLTE) – SERVICE IMPLEMENTATION & CELL PLANNING PERSPECTIVE

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Annotation. This chapter provides a short introduction to the Voice over Long Term Evolution (VoLTE). According to 3GPP release 8 LTE was introduced providing higher access rates and lower latency, more efficient use of radio network resources which means lower cost per transmitted bit and voice spectral efficiency. Circuit switched domain was excluded and all applications – services are implemented as packet switched services. On 2009 the One Voice alternative was published by a number of communication service providers and vendors. The conclusion was that the IMS-based solution as defined by 3GPP was the best way to meet the end users' expectations for service quality, availability, reliability when moving from existing circuit switched telephony services to IP-based LTE services. During 2010 the "One Voice" initiative was adopted by the Global System for Mobile Association (GSMA). That was supported by organizations, mobile service providers, vendors and handset manufacturers. The result of this action was a GSMA Voice over LTE (VoLTE) solution based on standards and supported by industry. Voice over LTE using IP based service poses several restrictions on Quality of Service, mainly over air interface. A general model approach is also presented in this chapter, contributing to the radio network designers planning algorithms and solutions.

1. Introduction

The Voice over LTE solution allows the operators to evolve from circuit switched (CS) based solution (Mobile Soft-switch solution, MSS) in WCDMA networks towards an IMS IP-based core network [1]. CS domain is not supported by LTE; consequently voice service is delivered as packet switched domain (PSD) through IP using the IP Multimedia Subsystem (IMS)-based standard. MSS and IMS use different switching nodes in the connectivity and control layer. However Mobile Media Gateway (M-MGw) can be used by MSS and IMS for transferring user data and signaling media payload according to 3GPP soft-switch layered architecture on connectivity network layer. MSS can handle voice calls via Wifi known as Generic Access Network (VOLGA) [2] and uses IP communication between UE and MSC server (MSC-S) being either over fixed or mobile broadband. MSS is also involved in Voice over LTE (VoLTE) when there is a need to roam between operators [3].

Media Gateway Control Function (MGCF), a standard switch node in the IP Multimedia Subsystem (IMS) which communicates with the Call Session Control Function (CSCF) and controls the connections for media channels in an IP Multimedia Subsystem Media Gateway (IMS-MGW). MGCF performs protocol conversion between ISDN User Part (ISUP) and the IMS call-control protocols. The MGCF can be embedded in MSC-S while the equipment needed for IMS connections is located in the firmware of MGW, known as IMS-MGW, being controlled by MGCF. IMS-MGW, is a component located in the IP Multimedia Subsystem (IMS) 3G architecture, which could terminate bearer channels from a circuit switched network and/or media streams from a packet network. It supports media conversion, bearer control, and payload processing (e.g., using codecs, echo cancellers, or conference bridges).

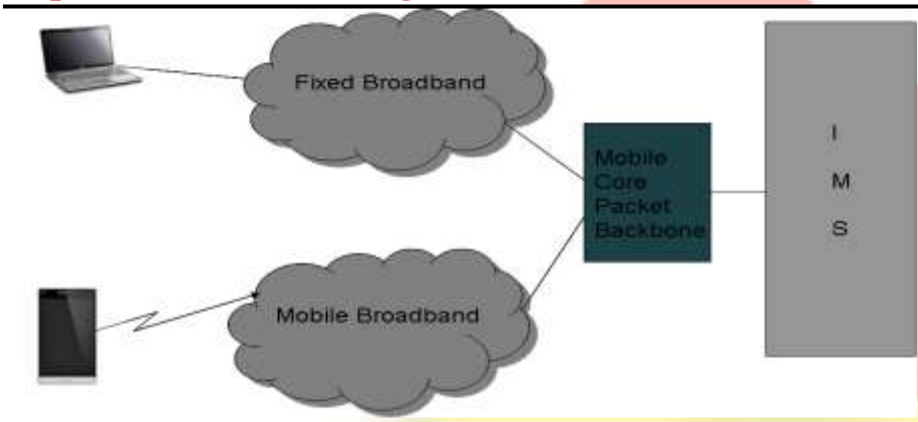

Fig. 1 Topology for LTE network supporting IMS

As a result we may summarize and say that IMS offers a standardized, future-proof network architecture with open interfaces, guaranteeing interoperability in multivendor environments and smooth evolution with maximum reuse of the existing networks. These advantages results to a smooth and operator decided evolutionary path from mobile soft switching (MMS) to IMS within the same Open Core system.

2. VoLTE using IMS overview

VoLTE [4] is based on multimedia telephony (MMTEL) service, a standardized IMS-based VoIP solution where Evolved Packet Core (EPC) domain provides access mobility connectivity and the IMS domain provides the call control functionality. Packet domain network Gateway (PGW) node establishes a diameter session towards Policy and Charging Rule Function (PCRF) over Gx interface. PCRF is mandatory for IMS voice sessions since bearer level QoS is based on the IMS voice media session information, negotiated for each voice call dynamically. For voice call PCRF also gets voice session information from the PCSCF and based on that information the PCRF decides the required QoS for a bearer to be established and this information is sent towards P-GW.

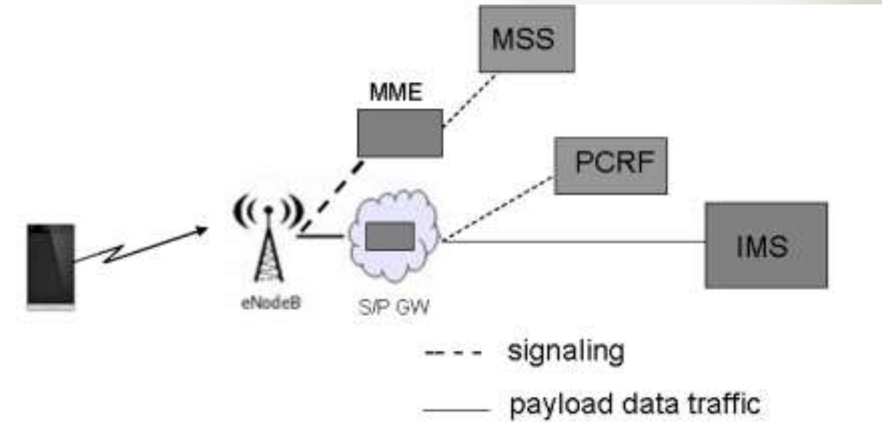

Fig. 2 Topology for IMS platform

LTE uses the concept of bearers to carry data between UE and core network and to provide Quality of Service (QoS) differentiation. QoS means that different services have different characteristics and demand different amount of data, bit rates and transport capabilities. The new dedicated bearer will have a Quality of service Class Identifier (QCI) value according to negotiated QoS in PCRF. This QCI is connected over radio domain to scheduling priorities on eNodeB. Session Management functionality in Mobility Management Entity (MME) establishes and handles connections between User Equipment (UE) and a Packet Data Network (PDN). Each PDN connection is assigned at least one bearer known as default bearer; depending on the service the connection is used for, a number of dedicated bearers is also established. Default bearer is a non-Guaranteed Bit Rate (GBR) bearer while dedicated bearer can be either a GBR or a non-GBR bearer. PGW also sends a proxy call session control function (P-CSCF) address in the PDN connectivity response that is also signaled to UE. An LTE user has at least one Packet Data Network (PDN) connection, when the user is registered into the EPS network. Specifically for VoLTE users, PDN connection is always established towards a well-known IMS Access Point Name (APN). APN will define the entry point into IMS domain. PDN Consequently "*connectivity request*" procedure is standardized to be used by UE, during IMSI attach procedure, to request the setup of a default EPS bearer to a PDN. P-GW allocates an IP address (Ipv4 or Ipv6) to be used by UE. The P-GW also sends a Proxy Call Session Control function (PCSCF) address in the PDN connectivity response that is signaled to the UE. When the PGW sends a request to PCRF it contains at least the UE IP address and radio access type. During PDN connection the PCRF may provide policy and charging control rules, for example the QCI to apply to a specific bearer.

Initial QoS settings of default bearer are assigned by the network based on subscription data. The number of dedicated established bearers can vary throughout the lifetime of the connection. A tunnel between UE and S/P GW for all bearers (default or dedicated) is assigned. According to 3GPP PDN connection is used to transport payload between the UE and the PDN, using one or more available EPS bearers. Information about the PDN connection is stored in the MME, according to figure 2, to be used when connections or bearers are for example modified or deactivated. According to standards and specifically for VoLTE solution default bearer is used for IMS signaling and has QCI = 5.

3. IMS multimedia Telephony (MMTel)

MMTel is an end-to-end network solution allowing operators to offer the same service over many different access types. It is a global standard based on the IMS, offering converged, fixed and mobile real-time multimedia communication using the media capabilities such as voice, real-time video, text, file transfer and sharing of pictures, audio and video clips. With MMTel, users are capable to add/drop media during a session. The user can start with chat, add voice (for instance Mobile VoIP), add another caller, add video, share media and transfer files, and drop any of these without losing or having to end the session. IMS/MMtel service consists of basic communication info and optional supplementary services. MMTel is prepared to handle voice calls from any UE supporting Session Initiated Protocol (SIP). Since there is a standardized Netqork to Network Interface (NNI) in MMTel, it is possible to interconnect all the multimedia features enabling operators to become world's largest multimedia community. In operator and vendor market perspective MMTel has been positioned as a future mass-market service for real-time multimedia communication. At launch time, about 2011-2012, initial MMTel service community was small; consequently to fit and be compatible with previous network topologies, it will have to interwork with existing mass-market services. 3GPP also has standardized interworking implementations between MMTel voice/ video and circuit-switched video telephony, as used in other 3GPP networks like WCDMA or GSM/GPRS.

From network planning point of view it is important to remember that 2G and 3G technologies evolve in parallel with 4G in existing operator networks. The consequence is that users can get their CS (voice, video) communication needs met by 2G and 3G instead of 4G, depending on how much bandwidth they demand and how much bandwidth is available due to other existing data services.

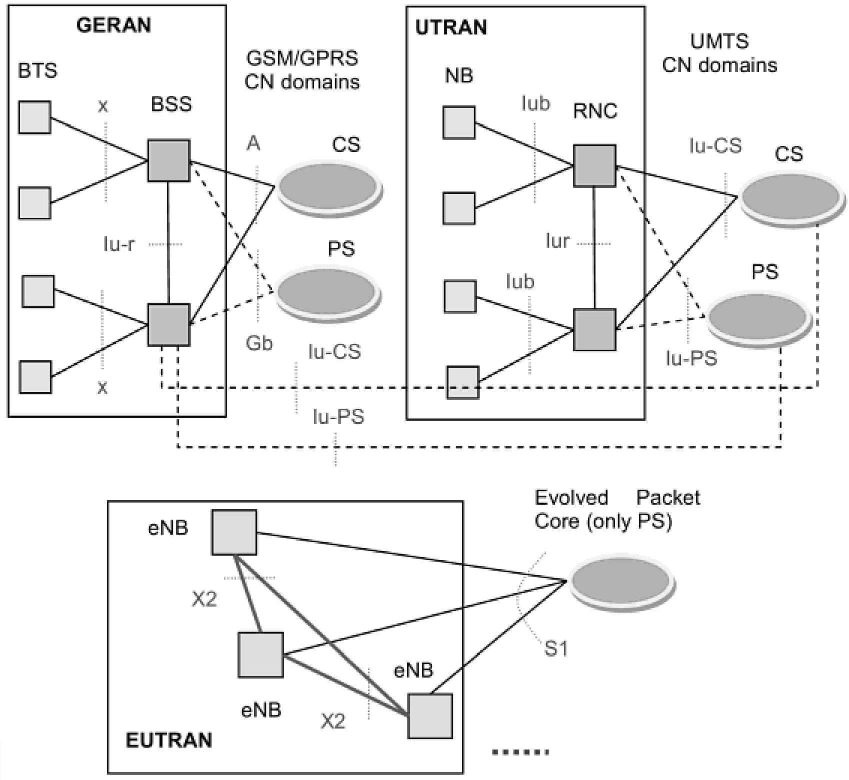

Fig. 3 Radio Access Networks Overview for cellular services

There are different options defining the implemented solution for a user, initially connected to LTE (VoLTE), to fallback to another technology 2G or 3G. Such options are:

SVLTE – Simultaneously 2G/3G Voice and 4G Data,

CSFB – CS Fallback (CSFB)

SRVCC – Single Radio Voice Call Continuity
3.1 SVLTE solutions
During SVLTE the handset works simultaneously in both LTE and CS mode.

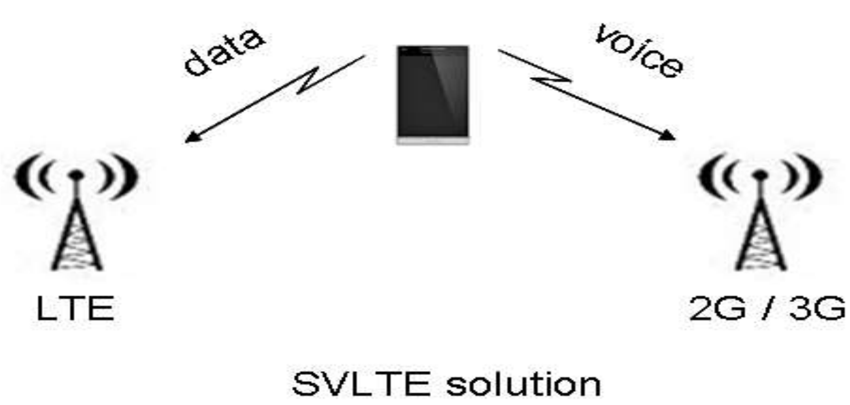

Fig. 4 LTE connectivity options in 2G/3G
The LTE mode provides data services and the CS mode providing voice services. SVLTE uses different antennas for CS connections over 2G/3G and PS connections over LTE. SVLTE is based on handset that will be able to support connection to both LTE and 2G/3G at the same time, and doesn't put any special requirements on the network. However these handsets could be expensive and consume more power.

3.2 CSFB solution

CSFB (Circuit Switched Fallback) is an interim solution in 3GPP Release 8. In this approach, the LTE provides data services, and when a voice call is to be initiated or received, it will fall back to the CS domain. When using this solution [5], operators need to upgrade the MSC instead of deploying the IMS, and therefore, can provide services quickly. However, the disadvantage is longer call setup delay.

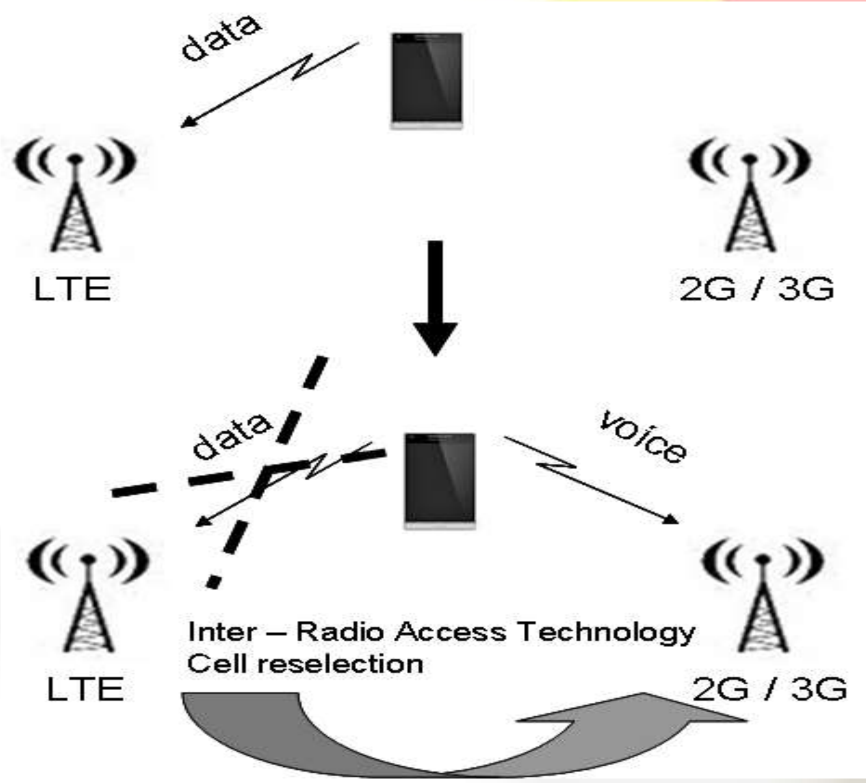

Fig. 5 LTE connectivity options in 2G/3G

There are however specified requirements to fall back to a CS voice service in a GSM or WCDMA network, if of course available in the same coverage area. CSFB requires that:

SGs interface between MME and MSC-S is defined,

the UE is dual radio capable

UE is registered in CS domain.

The MSC-S is updated and allow the CSFB to perfom paging via LTE.

Considering LTE implementation and operator budget restrictions, it is expected that LTE will initially provide coverage to small geographical areas with high user capacity, trying to absorb as much as possible high data traffic demands. Such areas are city centers, Malls, stadiums, business districts etc. In such areas service continuity must be provided when the subscriber is moving outside LTE's coverage area. In CSFB option the user connected to LTE will be redirected by the network to cell-reselection, from 4G to 2G (GERAN) or 3G (UTRAN) access network to connect to the CS domain, when there is a call request. The advantage of such network functionality performance is that CS Fallback might be used as a generic telephony fallback method securing functionality for incoming roamers as well. The registration, location update and paging procedures are the same for SMS and CSFB. From signaling messages and signaling needs, without explaining all details, CSFB performs following steps:

Subscriber is registered in MSC but roam in LTE

Incoming call to subscriber in LTE

MSC Paging over SGs, S1 and eUu-Interface

UE notifies MME with an Extended Service Request.

MME orders eNodeb to release and inform UE that CSFB is ok.

UE and RAN triggers an enhanced release with redirect

UE sends a location update and call setup over 2G/3G radio

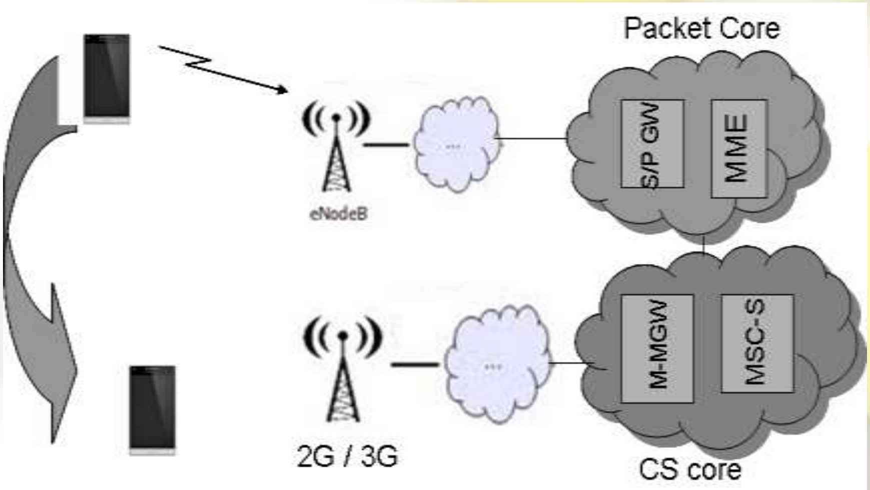

Fig. 6 LTE CSFB solution

3.3 VoLTE using MMTel

In order to provide call continuity and avoid any interruptions like IRAT cell changes or CS fallback transfer, VoLTE can perform handover transfer of ongoing voice call from LTE (EPS/IMS domain) to circuit switched domain in case native VoIP connection no longer can be maintained in LTE. This procedure is standardized under name Single Radio Voice Call Continuity (SRVCC) which means that only one radio technology apply at the same time in the UE and the services continue (handover), but is also known as packet switched –circuit switched (PS-CS) access domain transfer. SRVCC technology enables handover voice communication between VoLTE and CS in 3G. Enhanced SR-VCC (eSRVCC) is developed to facilitate the handover from VoLTE to circuit switch (CS) domain (GERAN/UTRAN).

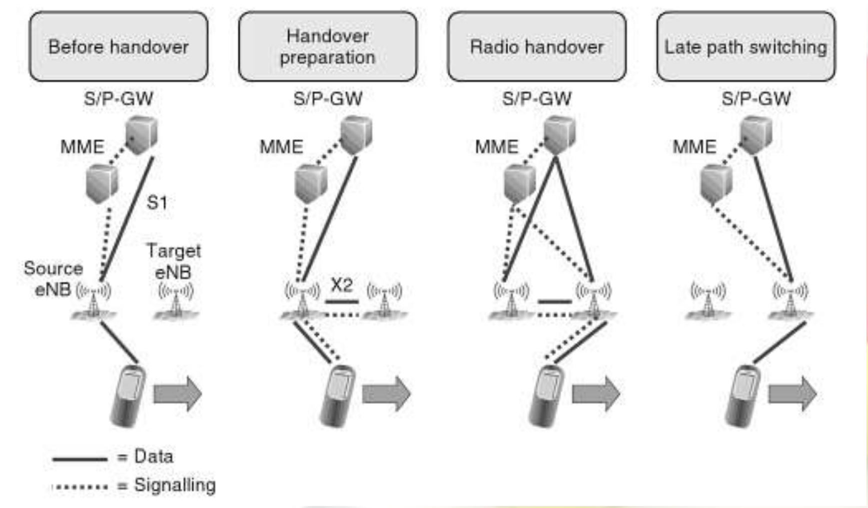

Fig. 7 LTE RAN mobility overview

In order to support access domain transfer new functionalities are required. A SRVCC enhanced MSC server that is able to perform required procedures towards circuit switched domain and also enhanced EPS and more specific MME and eNodeB. UEs will also have to specifically support the SRVCC function. It is important to remember that one of the main advantages of such a solution is that only Voice part of communication between UE and network is transferred from LTE to CS; other data bearers are maintained via the PS core.
Shortly describing, SRVCC enhanced MSC Server prepares resources towards IuCS or A interface for immediate domain transfer and updates remote side of session after domain transfer. After circuit switched resources have been committed the SRVCC enhanced MSC server will establish call on behalf of UE to a specific address given by MME over Sv interface.

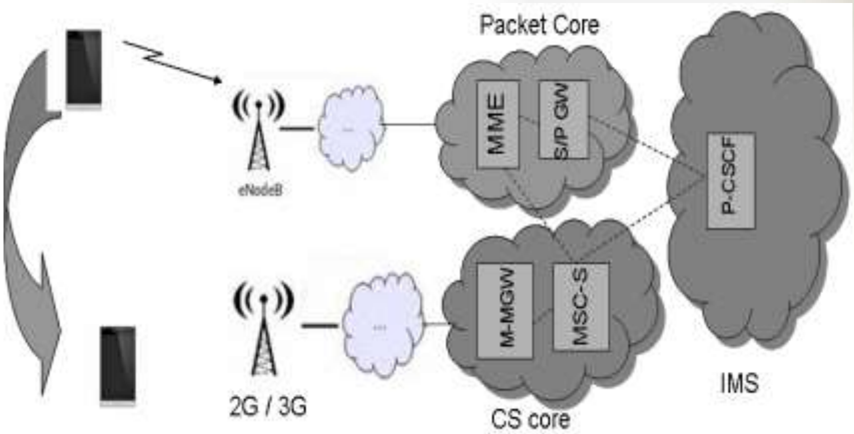

Fig. 8 SRVCC solution
When VoLTE user equipment (UE) makes an initial attachment it indicates its voice domain preference, (for example IMS PS voice over CS or only PS voice capable) and Single Radio Voice Call Continuity (SRVCC) capability. The MME uses the UE provided information, subscription information, local policy and SR-VCC capability of the network to decide if VoIP can be provided. The decision is signaled back to the UE in attach accept message. If VoIP can be supported the UE initiates IMS registration in order to initiate and receive VoIP communication otherwise UE will try CSFB to GSM or UTRAN. In order to facilitate the session transfer (SRVCC) of the voice component to the CS domain, the IMS MMTel sessions need to be anchored in the IMS. IMS asks MSC-S, via Packet Core, to take over the call. Services can still be provided from IMS, even though the subscriber is not handled by IMS. This is the purpose with IMS Centralized Services (ICS).
4. LTE Air Interface Overview
A new generation of wireless cellular networks, called Enhanced UTRAN (E-UTRAN) or Long Term Evolution (LTE) workgroup of 3GPP, has been evolved providing advantages to services and users over

broadband wireless [6, 21]. Advantages of LTE, compared to older broadband technologies, are strongly dependant on throughput and latency requirements. As an overall enhancement E-UTRAN should be able to support average data rates of up to 300 Mbps in downlink and 50 Mbps in the uplink. Considering also a 20 MHz FDD uplink and downlink spectrum allocation, it is expected to achieve 5 bps/Hz downlink spectrum efficiency and 2.5 bps/Hz uplink spectrum efficiency.

In typical LTE deployment, particularly in the eNodeB/cell dimensioning process, typical network requirements are coverage area, number of subscribers (also known in international literature as traffic load), traffic type (QoS reruirements from the core network), traffic model, transmission power and Uplink/Downlink cell edge throughput. Coverage area is the first parameter that a network planner is considering in order to be able to calculate the expected neighbour cell interference (also known as intercell interference) in the serving cell. Expected number of subscribers in the serving cell is also important factor since it provides a measure of how often users are scheduled and how often the common resources are used, contributing also to intracell interference factors. Traffic type (also known as service like VoIP) is also of important consideration since it provides a measure of the expected supported bit rate in the serving cell area of coverage. It is directly interconnected to the QoS of the supported cellular service and it is an input to the scheduler. LTE QoS complies with the 3GPP Rel 8 TS 23.203 QoS concept, providing priorities to different services. Different services are supported in 3GPP LTE and each one with a different QoS profile, however all of them based on IP. Traffic type and number of subscribers comprise an expected traffic load (in Erlangs) which influences, as an overall factor, the scheduling usage and the intercell interference. Transmission power is also affecting intercell interference; however it is not of such importance for the scheduling procedure. Finally in radio design process cell edge should mostly be studied since this is the geographical area with the lowest Signal-to-Interference & Noise ratio (SINR) factor, affecting thus scheduling and throughput. As a consequence radio air interface must be able to provide both high peak bit rates and acceptable cell-edge bit rates. However, besides the cell edge throughput requirements, during the deployment and dimensioning process important attention should be also given to the latency, specially for VoIP services, in order to provide an overall sufficient and satisfactory QoS.

3GPP LTE is based on OFDM principle over air interface. OFDM is in principle an efficient modulation scheme (and not a multiple access scheme as it is commonly referred in international literature) where each user has been allocated part of existing bandwidth (called sub-band) in specific time instances. OFDM principle divides allocated frequency band into a number of narrow 15 KHz sub-carriers. A minimum group of permitted subcarriers consists of 12 subcarriers of 180 KHz bandwidth. As modulation QPSK, 16QAM or 64QAM might be used depending on the channel conditions, thus representing different number of bits into OFDM symbols (QPSK 2 bits/symbol, 16QAM 4 bits/symbol and 64QAM 6 bits/symbol). Due to time-dispersive radio channels, where Inter-Symbol Interference (ISI) is present, a cyclic prefix as a time-offset is added to the OFDM symbol duration to maintain the time orthogonality between subcarriers, resulting into an OFDM symbol duration of $1/\Delta f$ + cyclic prefix. Each OFDM symbol is known as resource element. The transmission of information, meaning what resource a scheduler could schedule for transmission, is known as resource block. One resource blocks is a two-dimensional resource which has a total size of 180 KHz (12 sub-carriers of 15 KHz each) in the frequency domain and seven resource elements per sub-carrier of duration 0.5ms in the time domain, thus one resource block has 12 X 7 = 84 resource elements. Transmission over radio interface consists of two resource blocks known as 1ms Transmission Time Interval (TTI), as presented on figure 9. Allocated bandwidth consists of N X 180 KHz, is defined per sector and it could belong to a wider bandwidth which is allocated to the operator.

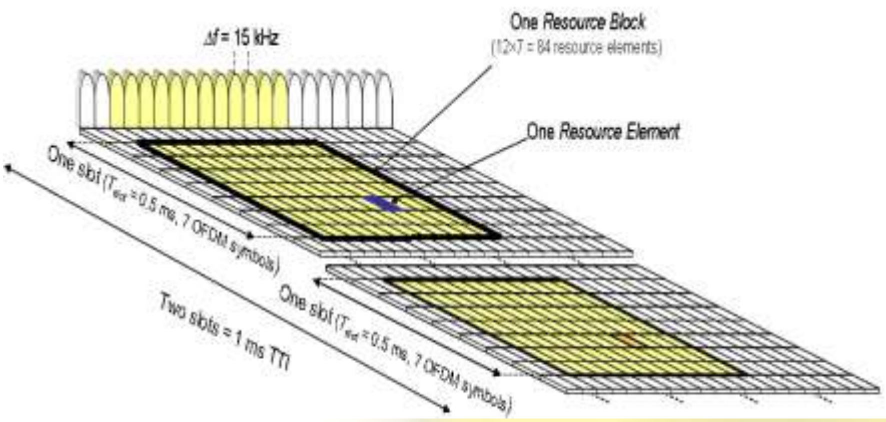

Fig. 9 OFDM principle – LTE air interface

If neighbouring cells are using same bandwidth or nearby spectrum the system is expected to be influenced strongly by interference, resulting into lower capacity (Shannon limit) and into lower throughput. Frequency reuse patterns and also more clever frequency handling techniques have been recently proposed and considered in international literature since they are connected directly to the SINR factor. LTE is indeed the evolution of High Speed Packet Access [HSPA] cellular networks towards 4G [6]. It provides backward compatibility to GSM-WCDMA existing networks and on the same time it is functionally compatible with WiMAX networks, easing the network planners for a broadband network heterogeneous planning convergence [7].

5. VoIP Quality of Servixe

One of the first considerations for cell planners is to optimize geographical coverage and provide adequate quality of service based on operator restrictions. Frequency reuse has been standardized by 3GPP [8-11] and also recently has been considered in international literature [12, 13]. MAC scheduler [20], being responsible for the dynamic allocation of frequency-time resources into many users, provides uplink decisions mainly based on:

signal to noise and interference ratio γ_{RB} measurements per resource block (RB),
required Quality of Service received from core network (Quality Class Identifier – QCI) [14]
cell load conditions (including interference and availability on RB)
Delay constraints

5.1 Service Requirements

To provide efficient resource usage for VoIP services, LTE concept supports fast scheduling [15] considering the importance of calculating radio delay mostly due to scheduling decisions. Following figure 10 according to 3GPP standards, VoIP delay is considered to be around 100 ms.

QCI	Resource Type	Priority	Packet Delay Budget	Packet Error Loss Rate	Example Services
1	GBR	2	100 ms	10^{-2}	Conversational Voice
2		4	150 ms	10^{-3}	Conversational Video (Live Streaming)
3		3	50 ms	10^{-3}	Real Time Gaming
4		5	300 ms	10^{-6}	Non-Conversational Video (Buffered Streaming)
5	Non-GBR	1	100 ms	10^{-6}	IMS Signalling
6		6	300 ms	10^{-6}	Video (Buffered Streaming), TCP-based (www, ftp, e-mail, chat, p2p file sharing, progressive video, etc.)
7		7	100 ms	10^{-3}	Voice, Video (Live Streaming) Interactive Gaming
8		8	300 ms	10^{-6}	Video (Buffered Streaming), TCP-based (www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9		9			

Fig. 10 3GPP QoS standards per QCI and per service – VoIP is GBR priority 2

5.2 Cell Planning Process for VoIP services

Cell planning process is important to include VoIP delay constraints. Following steps should be considered:

Pathloss estimation

Initially consider the operator selected cell range as initial important coverage constraint. Mainly due to operator determined restrictions regarding user estimated throughput, pathloss $L_{celledge}$ at worst radio conditions (cell edge user for outdoor planning) have to be estimated. Estimations should be based on certain defined pathloss models, where a well defined formula for 2.5 GHz LTE microcell outdoor to outdoor coverage is [16]:

$$L[dB] = \begin{cases} 39+20\log_{10}(d[m]), & 10m<d\leq 45m \\ -39+67\log_{10}(d[m]), & d>45m \end{cases} \quad (1)$$

The noise floor per RB N_{RB} has to be calculated.

Noise N_{RB} per resource block is considered to be -174 dB/Hz and for 180 kHz resource block bandwidth it is calculated as -111.44 dB [17].

Intercell Interference

At worst cell conditions for outdoor planning (cell edge user) uplink Interference per RB has to be calculated. Interference is mainly considered to be inter-cell interference from a neighbour cell. From cell planner perspective we do consider that it is more accurate to have an average estimation of inter cell interference per resource block, at a given path loss, from real drive test measurements. Appropriate plots of Absolute Interference per RB vs. cell edge Path Loss L_{target} have been created from drive test according to Fig. 11.

Uplink Signal to Noise and Interference ratio γ at cell edge

Uplink γ ratio is extremely important to be estimated since it is directly related to MAC scheduler link adaptation [20], affecting the RB selection on uplink scheduling. Of course on cell edge conditions (worst conditions for outdoor planning) the target γ has to be always higher than the eNB receiver sensitivity S_{eNodeB}, which is defined as the minimum received power on RBS required to correctly decode uplink RB with $1x10^{-10}$ bit error rate [17]:

$$S_{eNodeB} = N_T \cdot N_{fig} \cdot B \cdot \gamma_{target} \quad (2)$$

On equation (2) N_T is the thermal noise power density calculated, from Boltzmann's constant $k_B = 1.38 \times 10^{-23}$ $J/^0K$ and the absolute temperature in Kelvin $T = 290\ ^0K$, to be -174 dB/Hz. Moreover N_{fig} is the eNodeB noise figure defining a degradation of γ due to RF circuitry components, calculated to be 2 dB for uplink [17, 18]. Finally B is the resource block bandwidth of 180kHz .Substituting variables into (2) we get $S_{eNodeB} = -104.5 + \gamma_{target}$ dB. Proceeding with figure 11 and considering also an operator predefined path loss at cell edge, γ_{target} could be properly estimated [19, 20]. M_{LNF} is the log-normal fading margin given by Jakes formula for a certain percentage of coverage and for specific environment (Urban, Dense Urban, Suburb etc.). L_{BL} is the expected body loss considered either as 2 dB for handset palm-top or 0 dB for lap-top:

$$\begin{aligned} L_{oper} &= P_{T,s}^{UE,RB} - S_{eNodeB} - M_{LNF} - L_{BL} \Rightarrow \\ \gamma_{target}[dB] &= 144.45 - L_{oper} - M_{LNF} - L_{BL} \end{aligned} \quad (3)$$

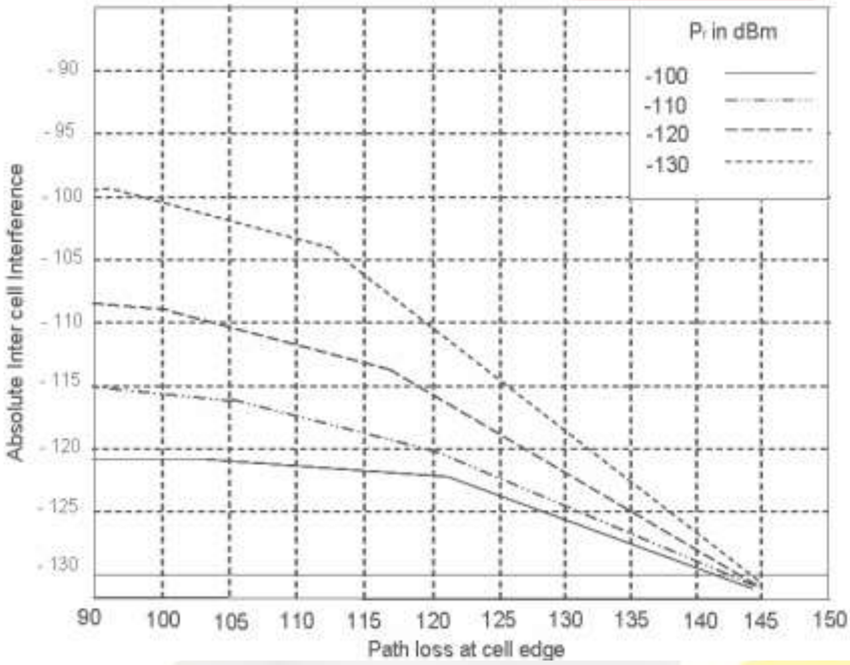

Fig. 11 Interference estimation vs.pathloss - drive tests results
Estimate scheduler average number of uplink allocated RBs n_{RB}

Based on the target γ_{target} on cell edge, the number of allocated resource blocks n_{RB} is calculated considering uniform power distribution of nominal UE power P_{UE} over all transmitted resource blocks, as presented in figure 12.

$$\gamma_{target} = \frac{P_{UE}^{RB}}{noise+interference} = \frac{P_{UE}^{RB} / \left(L_{oper} \cdot n_{RB}\right)}{\left(N_{RB}+I_{RB}\right)} \Rightarrow \tag{4}$$

$$n_{RB} = \frac{P_{UE}^{RB}}{L_{oper} \cdot \left(N_{RB}+I_{RB}\right) \cdot \gamma_{target}}$$

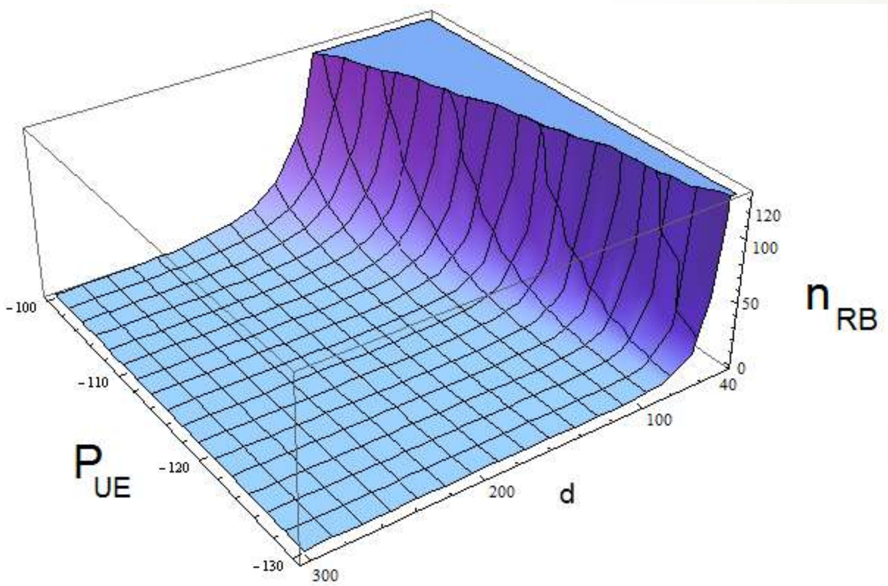

Fig. 12 number of resource blocks vs. user equipment uplink power and distance to eNodeB.
Estimate transmission rate per RB.

To estimate the expected transmission rate per RB vs. existing signal to noise ration γ, real drive test measurements have been analyzed. Since radio channel conditions are related to user velocity due to Doppler fadings, drive tests and the appropriate analysis have been executed for three different environments; EPA5 for pedestrians with average velocity of 5 km/h, EVA70 for in car driving users with average velocity of 70 km/h, ETU300 for high speed users on highways [17]. Expected curves are presented on figure 13.

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