Chapter 3
Voice Over LTE (VoLTE): Service Implementation and Cell Planning Perspective

Spiros Louvros and Angelina Gkioni

Abstract This chapter provides a short introduction to the voice over long-term evolution (VoLTE). According to 3G Patent Platform (3GPP) release 8, LTE was introduced providing higher access rates and lower latency and 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. In 2009, the One Voice alternative was published by a number of communication service providers and vendors. The conclusion was that the IP multimedia subsystem (IMS)-based solution as defined by 3GPP was the best way to meet the end users' expectations for service quality, availability, and 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 VoLTE solution based on standards and supported by industry. VoLTE using IP-based service poses several restrictions on quality of service (QoS), mainly over air interface. A general model approach is also presented in this chapter, contributing to the radio network designers planning algorithms and solutions.

3.1 Introduction

The voice over long-term evolution (VoLTE) solution allows the operators to evolve from circuit-switched (CS)-based solution (mobile soft-switch solution (MSS)) in wideband code division multiple access (WCDMA) networks toward an IP multimedia subsystem (IMS)- based core network [1]. CS domain is not supported by LTE; consequently, voice service is delivered as packet-switched domain (PSD) through

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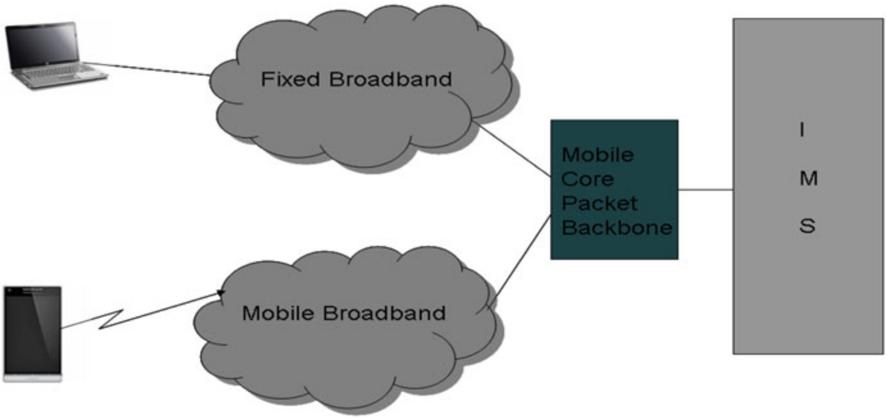

Fig. 3.1 Topology for long-term evolution (LTE) network supporting IP multimedia subsystem (IMS)

IP using the 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 3G Patent Platform (3GPP) soft-switch-layered architecture on connectivity network layer. MSS can handle voice calls via Wi-Fi known as generic access network (VOLGA) [2] and uses IP communication between user equipment (UE) and mobile switching center server (MSC-S) being either overfixed or mobile broadband. MSS is also involved in VoLTE when there is a need to roam between operators [3].

Media gateway control function (MGCF), a standard switch node in the IMS, which communicates with the call session control function (CSCF) and controls the connections for media channels in an IMS 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 whereas 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 IMS 3G architecture, which could terminate bearer channels from a CS 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. 3.1).

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 result in a smooth and operator-decided evolutionary path from mobile soft switching (MMS) to IMS within the same Open-Core system.

3.2 VoLTE Using IMS Overview

VoLTE [4] is based on multimedia telephony (MMTel) service, a standardized IMS-based voice over Internet Protocol (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 toward policy and charging rule function (PCRF) over Gx interface. PCRF is mandatory for IMS voice sessions since bearer-level quality of service (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 toward PGW.

LTE uses the concept of bearers to carry data between UE and core network and to provide 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 QoS 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 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 nonguaranteed bit rate (non-GBR) bearer whereas dedicated bearer can be either a GBR or a non-GBR bearer. PGW also sends a proxy CSCF (P-CSCF) address in the PDN connectivity response that is also signaled to UE. An LTE user has at least one PDN connection, when the user is registered into the evolved packet system (EPS) network. Specifically for VoLTE users, PDN connection is always established toward 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 international mobile subscriber identity (IMSI) attach procedure, to request the setup of a default EPS bearer to a PDN. PGW allocates an IP address (Ipv4 or Ipv6) to be used by UE. The PGW also sends a P-CSCF 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 Fig. 3.2, 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 $\mathrm{QCI} = 5$.

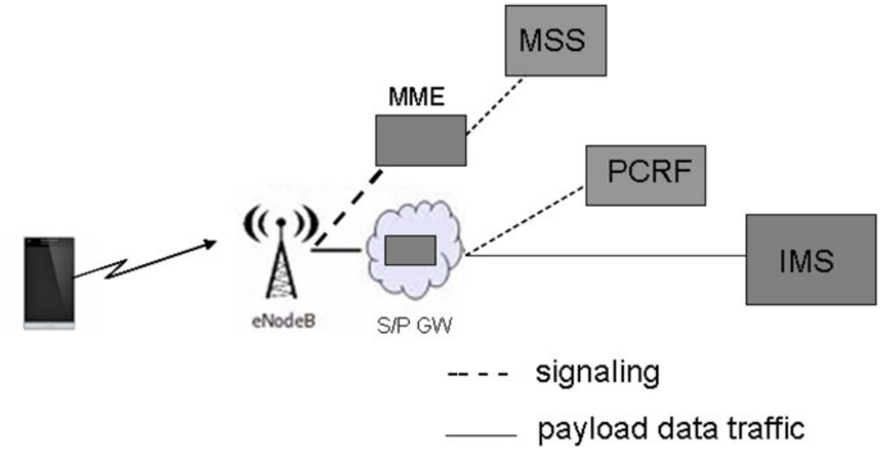

Fig. 3.2 Topology for IP multimedia subsystem (IMS) platform

3.3 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 information and optional supplementary services. MMTel is prepared to handle voice calls from any UE supporting session-initiated protocol (SIP). Since there is a standardized network-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 perspectives, MMTel has been positioned as a future mass-market service for real-time multimedia communication. At launch time, around 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. The 3GPP has also standardized interworking implementations between MMTel voice/video and circuit-switched video telephony, as used in other 3GPP networks such as WCDMA or global system for mobile communications (GSM)/general packet radio service (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

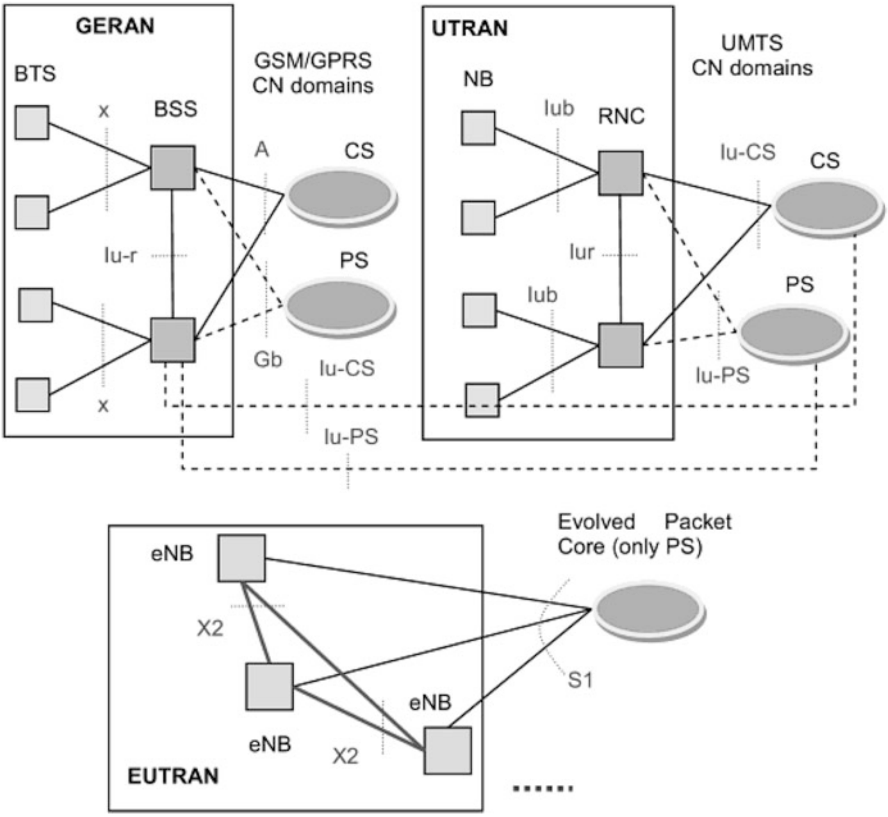

Fig. 3.3 Radio access networks (RANs) overview for cellular services

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. 3.3).

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:

- Simultaneously 2G/3G voice and 4G data (SVLTE).
- CS fallback (CSFB).
- Single radio voice call continuity (SRVCC).

3.3.1 SVLTE Solutions

During SVLTE, the handset works simultaneously in both LTE and CS mode (Fig. 3.4).

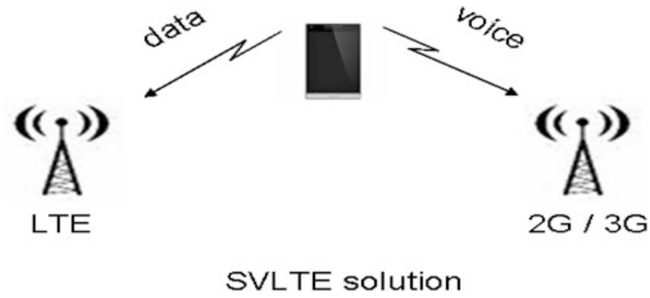

Fig. 3.4 Long-term evolution (LTE) connectivity options in 2G/3G

The LTE mode provides data services and the CS mode provides voice services. SVLTE uses different antennas for CS connections over 2G/3G and packet-switched (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 does not put any special requirements on the network. However, these handsets could be expensive and consume more power.

3.3.2 CSFB Solution

CSFB 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 falls 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. 3.5).

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 allows the CSFB to perform 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 high data traffic demands as much as possible. 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 CSFB might be used as a generic telephony fallback

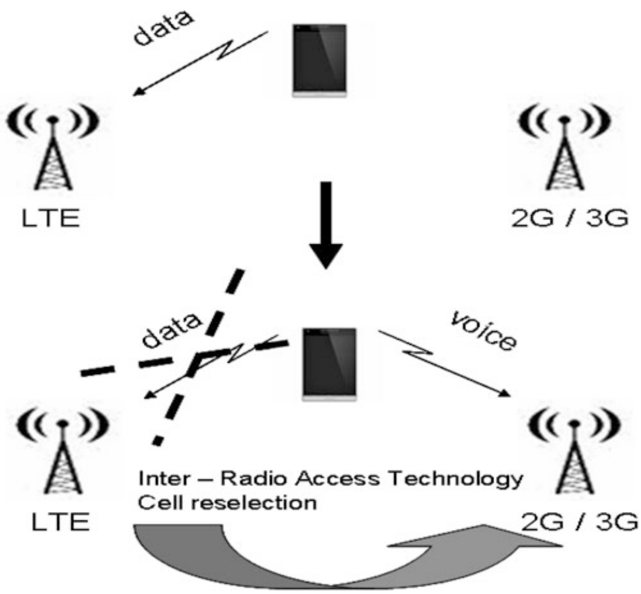

Fig. 3.5 Long-term evolution (LTE) connectivity options in 2G/3G

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 the following steps:

1. Subscriber is registered in MSC, but roams in LTE.
2. Incoming call to subscriber in LTE.
3. MSC paging over SGs, S1, and enhanced Uu 3GPP (eUu) interface (Uu interface is the radio interface between the mobile and the radio access network eNodeB).
4. UE notifies MME with an extended service request.
5. MME orders eNodeB to release and inform UE that CSFB is ok.
6. UE and RAN trigger an enhanced release with redirect.
7. UE sends a location update and call setup over 2G/3G radio (Fig. 3.6).

3.3.3 VoLTE Using MMTel

In order to provide call continuity and avoid any interruptions such as interradio access technology (IRAT) cell changes or CSFB transfer, VoLTE can perform handover transfer of ongoing voice call from LTE (EPS/IMS domain) to CS domain in case native VoIP connection can no longer be maintained in LTE. This procedure is standardized as SRVCC, which means that only one radio technology applies at the same time in the UE and the services continue (handover), but is

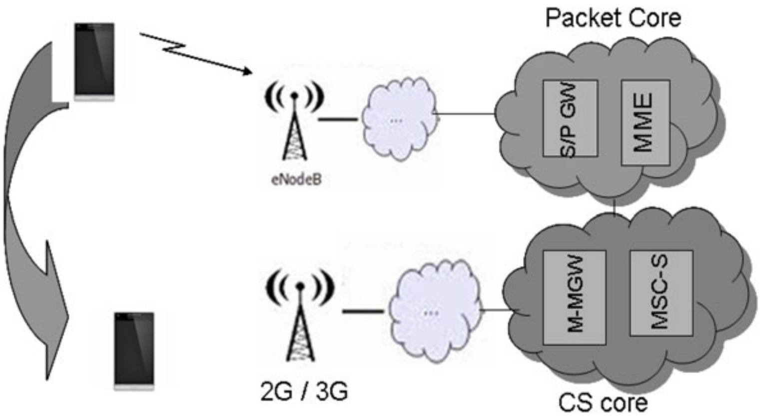

Fig. 3.6 Long-term evolution (LTE) circuit-switched fallback (CSFB) solution

also known as PS–CS access domain transfer. SRVCC technology enables handover voice communication between VoLTE and CS in 3G. Enhanced SRVCC (eSRVCC) is developed to facilitate the handover from VoLTE to CS domain (GERAN/UTRAN; Fig. 3.7).

In order to support access domain transfer, new functionalities are required. An SRVCC-enhanced MSC-S that is able to perform required procedures toward CS 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-S prepares resources toward IuCS or an interface for immediate domain transfer and updates remote side of session after domain transfer. After CS resources have been committed, the SRVCC-enhanced MSC-S will establish call on behalf of UE to a specific address given by MME over Sv interface (Fig. 3.8).

When VoLTE UE makes *an initial attachment*, it indicates its voice domain preference (e.g., IMS PS voice over CS or only PS voice-capable) and SRVCC capability. The MME uses the UE-provided information, subscription information, local policy, and SRVCC 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 of IMS centralized services (ICS).

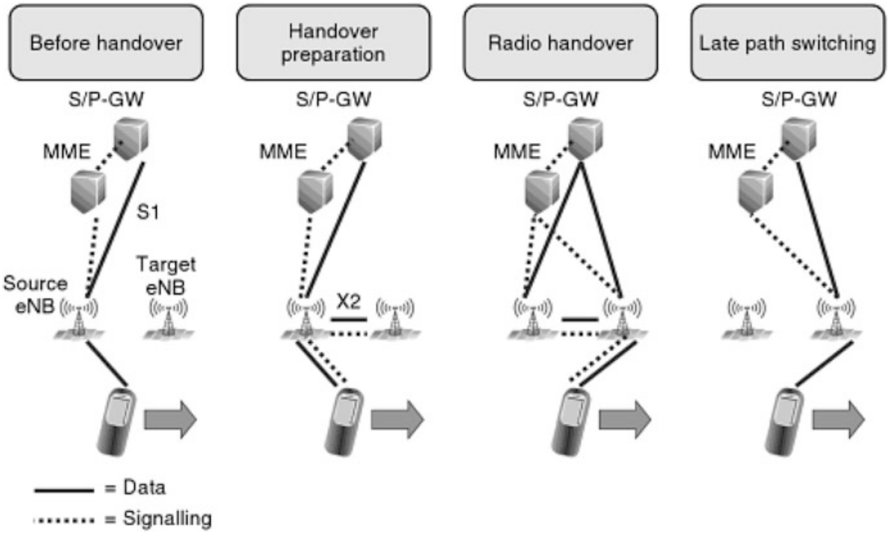

Fig. 3.7 Long-term evolution (LTE) radio access network (RAN) mobility overview

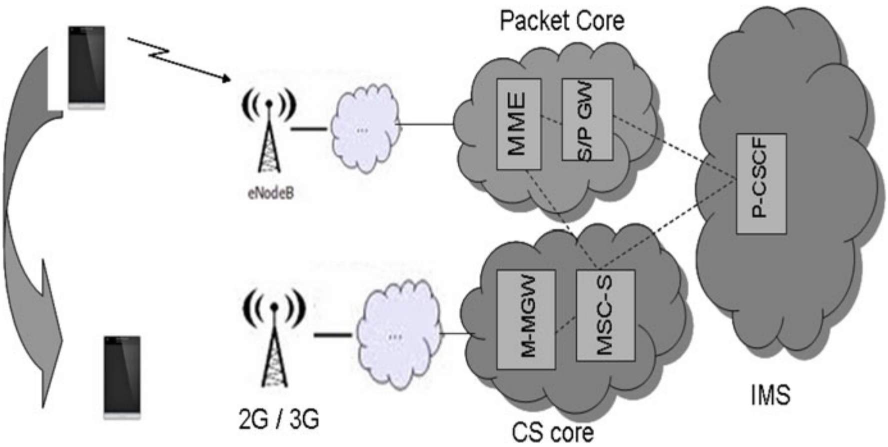

Fig. 3.8 Single radio voice call continuity (SRVCC) solution

3.4 LTE Air Interface Overview

A new generation of wireless cellular networks, called enhanced UTRAN (E-UTRAN) or LTE workgroup of 3GPP, has been evolved providing advantages to services and users over broadband wireless [6, 21]. Advantages of LTE, compared with 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 the 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 requirements 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 neighbor 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, also contributing to intercell 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 are 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 as this is the geographical area with the lowest signal-to-interference and noise ratio (SINR) factor, thus affecting 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, especially for VoIP services, in order to provide an overall sufficient and satisfactory QoS.

The 3GPP LTE is based on orthogonal frequency-division multiplexing (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 subband) in specific time instances. OFDM principle divides allocated frequency band into a number of narrow 15 kHz subcarriers. A minimum group of permitted subcarriers consists of 12 subcarriers of 180 kHz bandwidth. As modulation quadrature phase shift keying (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 intersymbol 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

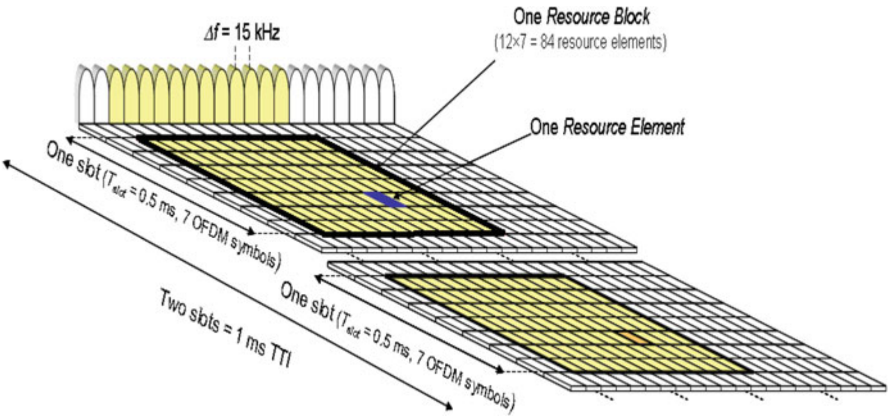

Fig. 3.9 Orthogonal frequency-division multiplexing (OFDM) principle: long-term evolution (LTE) air interface

for transmission, is known as resource block (RB). One RB is a two-dimensional resource, which has a total size of 180 kHz (12 subcarriers of 15 kHz each) in the frequency domain and seven resource elements (RE) per subcarrier of duration 0.5 ms in the time domain, thus one RB has $12 \times 7 = 84$ RE. Transmission over radio interface consists of two RBs known as 1 ms transmission time interval (TTI), as presented in Fig. 3.9. Allocated bandwidth consists of $N \times 180$ kHz, is defined per sector and it could belong to a wider bandwidth, which is allocated to the operator.

If neighboring cells are using same bandwidth or nearby spectrum, the system is expected to be strongly influenced 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 toward 4G [6]. It provides backward compatibility to GSM-WCDMA existing networks and at the same time it is functionally compatible with WiMAX networks, easing the network planners for a broadband network heterogeneous planning convergence [7].

3.5 VoIP Quality of Service

One of the first considerations for cell planners is to optimize geographical coverage and provide adequate QoS 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

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. 3.10 3GPP quality of service (QoS) standards per QoS class identifier (QCI) and per service: voice over Internet Protocol (VoIP) is guaranteed bit rate (GBR) priority 2

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 RB.
- Required QoS received from core network (QCI) [14].
- Cell load conditions (including interference and availability on RB).
- Delay constraints.

3.5.1 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 Fig. 3.10 according to 3GPP standards, VoIP delay is considered to be around 100 ms.

3.5.2 Cell Planning Process for VoIP Services

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

- Path loss estimation.
  Initially, consider the operator-selected cell range as initial important coverage constraint. Mainly due to operator-determined restrictions regarding user-

estimated throughput, path loss $L_{celledge}$ at worst radio conditions (cell-edge user for outdoor planning) have to be estimated. Estimations should be based on certain defined path loss models, where a well-defined formula for 2.5 GHz LTE microcell outdoor-to-outdoor coverage is [16]:

$$L[\mathrm{d}B] = \begin{Bmatrix} 39 + 20\log_{10}(d[\mathrm{m}]), & 10\,m < d \leq 45\ \mathrm{m} \\ -39 + 67\log_{10}(d[\mathrm{m}]), & d > 45\ \mathrm{m} \end{Bmatrix} \tag{3.1}$$

- The noise floor per RB N_{RB} has to be calculated.
  Noise N_{RB} per RB is considered to be − 174 dB/Hz and for 180 kHz RB 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 intercell interference from a neighbor cell. From cell planner perspective, we do consider that it is more accurate to have an average estimation of intercell interference per RB, at a given path loss, from real-drive test measurements. Appropriate plots of absolute interference per RB versus cell-edge Path Loss L_{target} have been created from drive test according to Fig. 3.11.
- Uplink signal-to-noise and interference ratio γ at cell edge.
  Uplink γ ratio is extremely important to be estimated as 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 1×10^{-10} bit error rate (BER) [17]:

$$S_{eNodeB} = N_T \cdot N_{fig} \cdot B \cdot \gamma_{target} \tag{3.2}$$

In Eq. 3.2, N_T is the thermal noise power density calculated from Boltzmann's constant $k_B = 1.38 \times 10^{-23}$ J/K and the absolute temperature in Kelvin $T = 290$ K, 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 RB bandwidth of 180 kHz. Substituting variables into Eq. (3.2), we get $S_{eNodeB} = -104.5 + \gamma_{target}$ dB. Proceeding with Fig. 3.11 and also considering 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 palmtop or 0 dB for laptop:

$$L_{oper} = P_{T,s}^{UE,RB} - S_{eNodeB} - M_{LNF} - L_{BL} \Rightarrow$$
$$\gamma_{target}[dB] = 144.45 - L_{oper} - M_{LNF} - L_{BL} \tag{3.3}$$

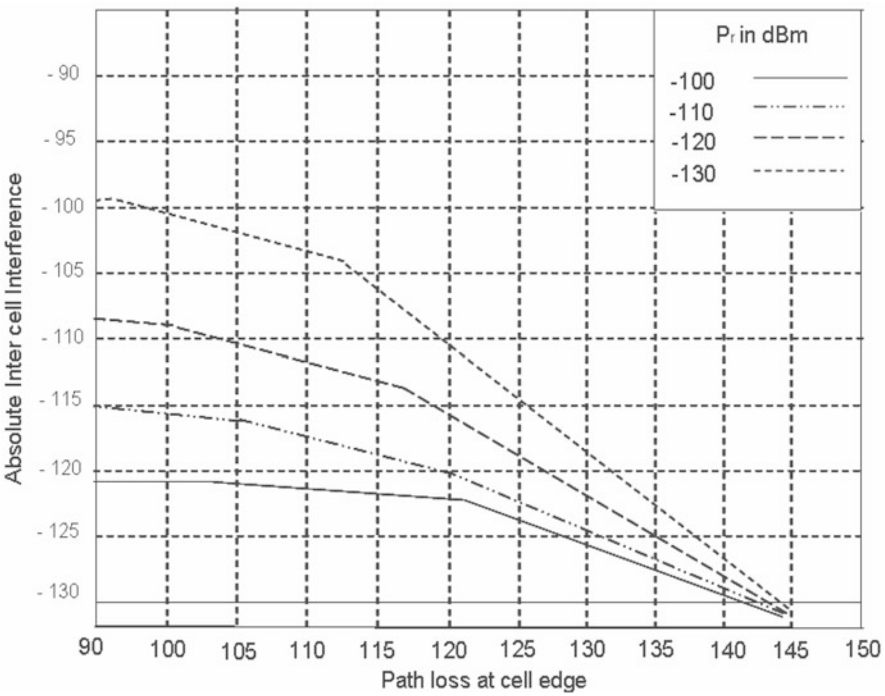

Fig. 3.11 Interference estimation versus path loss: drive tests results

- Estimate scheduler average number of uplink allocated RBs n_{RB}.
  Based on the target γ_{target} on cell edge, the number of allocated RBs n_{RB} is calculated considering uniform power distribution of nominal UE power P_{UE} over all transmitted RBs, as presented in Fig. 3.12.

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

$$n_{RB} = \frac{P_{UE}^{RB}}{L_{oper} \cdot (N_{RB} + I_{RB}) \cdot \gamma_{target}} \tag{3.4}$$

- Estimate transmission rate per RB.
  To estimate the expected transmission rate per RB versus existing signal-to-noise ratio 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, and ETU300 for high-speed users on highways [17]. Expected curves are presented in Fig. 3.13.

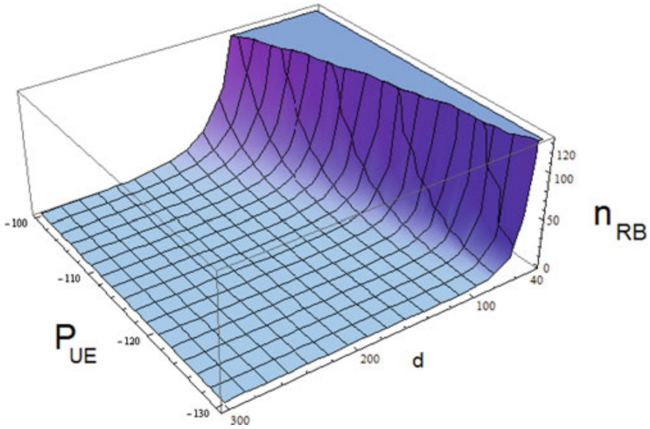

Fig. 3.12 Number of resource blocks (RBs) versus user equipment uplink power and distance to eNodeB

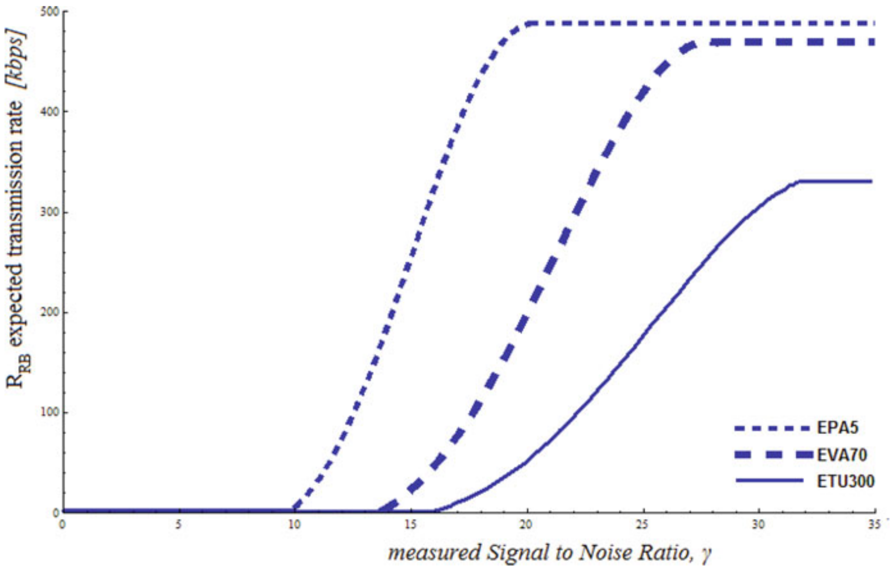

Fig. 3.13 Expected transmission rate per RB versus signal-to-noise ratio γ

- Calculate the expected total cell-edge transmission rate.
  Expected total transmission rate on cell edge is estimated by multiplying the average allocated number of resources n_{RB} from Eq. (3.4) with the expected transmission rate per RB from Fig. 3.13.

$$\langle R_{celledge} \rangle = n_{RB} \cdot R_{RB} \tag{3.5}$$

Average expected rate for VoIP service should be around 20 kbps, also considering retransmissions.

- Estimate expected retransmission rate
  The average number of retransmissions m is a function of the expected packet error rate over physical OFDM layer. One MAC packet is considered to be corrupted if the BER is above a certain threshold; such a packet is retransmitted maximum v times before it is discarded and requesting retransmission from upper layers (RLC). Setting p to be the packet nonsuccessful probability (error probability) and assuming that this probability is small, the mean number of retransmissions can be calculated as:

$$m = p^v + \sum_{k=0}^{v-1} (k+1)p^k(1-p) = \frac{1-p^v}{1-p} \approx \frac{1}{1-p}, \quad p \ll 1 \tag{3.6}$$

  Nonsuccessful probability is a function of MAC packet length M_{mac} and bit error probability p_b according to [16] is:

$$p = 1 - (1 - p_b)^{M_{mac}} \tag{3.7}$$

  The average number of retransmissions is approximated as:

$$m \approx \frac{1}{1-p} = (1-p_b)^{-M_{mac}} \approx (1 + M_{mac} \cdot p_b), \quad p_b \ll 1 \tag{3.8}$$

  To estimate the expected bit error probability p_b, drive tests have been performed and the expected BER is calculated versus bit energy per noise spectral density, as presented in Fig. 3.14. The average size of MAC packet M_{mac} is difficult to estimate since its length is variable and explicitly decided upon MAC link adaptation, transport format selection functionality, and service, respectively. To proceed with drive tests, it is necessary to use network statistics from eNB. As an example, Ericsson eNodeB provides several statistical counters that could estimate the average packet length. Ericsson counter *pmUeThpVolUl [kb]* measures uplink MAC SDU volume and finally Ericsson counter *pmUeThpTimeUl [ms]* provides the period of MAC volume measurements in milliseconds, hence *pmUeThpVolUl/pmUeThpTimeUl*bits/1 ms provides average M_{mac}-transmitted bits per TTI.

- Estimate expected cell-edge throughput.
  Expected throughput is easily calculated from Eqs. (3.5) and (3.8) as:

$$\langle T_{celledge} \rangle = \langle R_{celledge} \rangle \cdot \left(1 - \frac{m}{100}\right) = n_{RB} \cdot R_{RB} \cdot \left(1 - \frac{m}{100}\right) \tag{3.9}$$

- Estimate the average transmission delay.
  To proceed with the expected delay, it is important to split the packet transmission delay into two parts. Consider the load $\rho = \lambda/\mu$ of the ratio of packet arrivals over

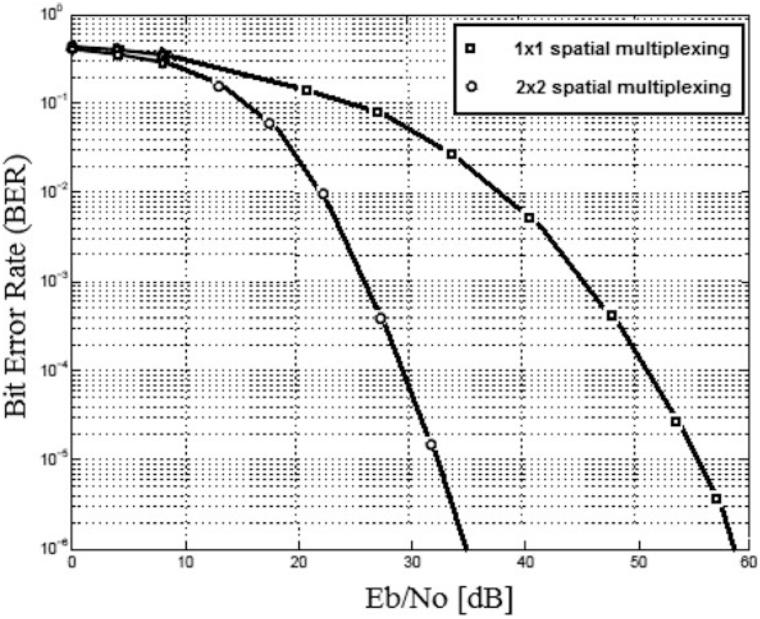

Fig. 3.14 Expected bit error rate (BER) versus bit received energy per noise spectral density

packet service time and in the general case suppose that specifically n packets exist in the system. Also, the signal processing delay per packet should be less or equal to one TTI = 1 ms. During first phase, a model for service layer IP source packet arrivals on UE, as they flow into the RLC/MAC layer waiting to be scheduled, is estimated [17] and presented in Fig. 3.15.

$$\begin{aligned}\bar{W} &= \sum_{n=1}^{\infty} n\pi_n \cdot TTI = \\ &\sum_{n=1}^{\infty} n\left[(1-\rho)\sum_{k=1}^{n}\left\{(-1)^{n-k}e^{k\rho}\left[\frac{(k\rho)^{n-k}}{(n-k)!}\right]\right\}\right]\cdot TTI + \\ &\sum_{n=1}^{\infty} n\left[(1-\rho)\sum_{\substack{k=1\\k\neq n}}^{n}\left\{(-1)^{n-k}e^{k\rho}\left[\frac{(k\rho)^{n-k-1}}{(n-k-1)!}\right]\right\}\right]\cdot TTI\end{aligned} \tag{3.10}$$

Second phase considers the MAC layer transmission, after uplink scheduling, where IP packets are forwarded into RLC/MAC layer for transmission over the physical medium. As expected from 3GPP standards and LTE protocols, IP packets will be segmented into many RLC/MAC-signaling data units (SDUs) to be mapped into OFDM RBs and transmitted over air interface. Multiple consecutive RBs n_{RB}

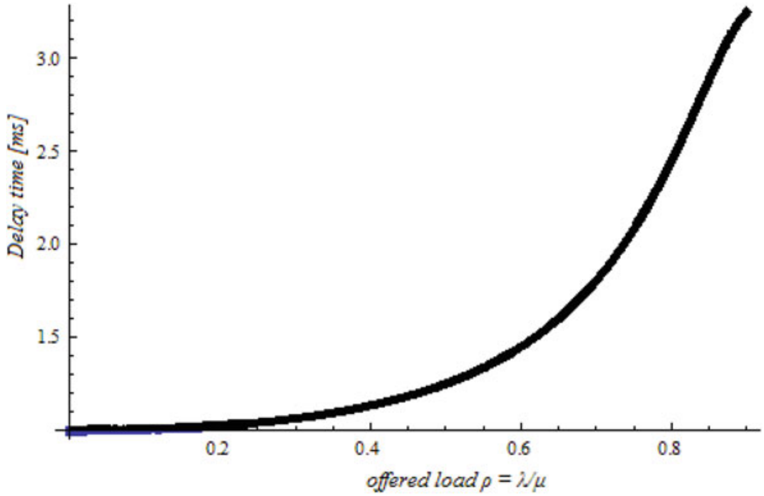

Fig. 3.15 Expected delay for service layer IP packet arrival on UE microprocessor

might be granted from scheduler for uplink transmission, minimizing the transmission latency and improving the UE throughput. Expected analysis will be based on transmissions of IP packets over RLC/MAC blocks based on channel conditions [16]. For a specific IP packet of average length M_I, the process defines fragmentation into $M_I + \lceil M_I/M_{mac} \rceil$ total number of RLC/MAC packets with $M_I + \lceil M_I/M_{mac} \rceil \cdot M_{over}$ total number of transmitted bits with fixed number of M_{over} header bits of 20 bytes per packet [20]. Considering also nonideal radio channel conditions, in such a scenario, the transmission time needed to completely transmit the IP packet will increase due to eventual retransmissions and nonscheduling periods of time. In general case, consider n_t number of uplink transmitted bits per RB (depending on MAC scheduler Link Adaptation Modulation and Coding Scheme), n_{RB} average allocated number of 180 kHz RB blocks per T_s transmission interval (as estimated in previous step), n_{AP} spatial multiplexing rank (MIMO or Tx diversity) and finally n and m integers indicating the average number of T_s units of time one MAC packet is not scheduled by scheduler and the average number of retransmissions one packet should undergo due to channel conditions, respectively (as calculated in previous step). Expected average whole IP packet transmission time would be:

$$W_{mac} = \frac{M_I + \lceil M_I/M_{mac} \rceil \cdot M_{over}}{n_{AP} \cdot n_{RB} \cdot n_t} \cdot TTI + (m+n) \cdot TTI \tag{3.11}$$

Uplink parameters n_t and n_{AP} could be easily calculated for cell-edge UEs considering that MAC scheduler will allocate QPSK modulation (2 bits per symbol) with Tx diversity, thus $n_{AP} = 1$. One subframe contains $14 \times 12 = 168$ REs where two OFMD symbols (24 REs) are devoted for sounding reference signals. Following Fig. 3.16, available number of uplink transmitted bits will be $n_{Ts} = (168 - 24) \times 2 =$

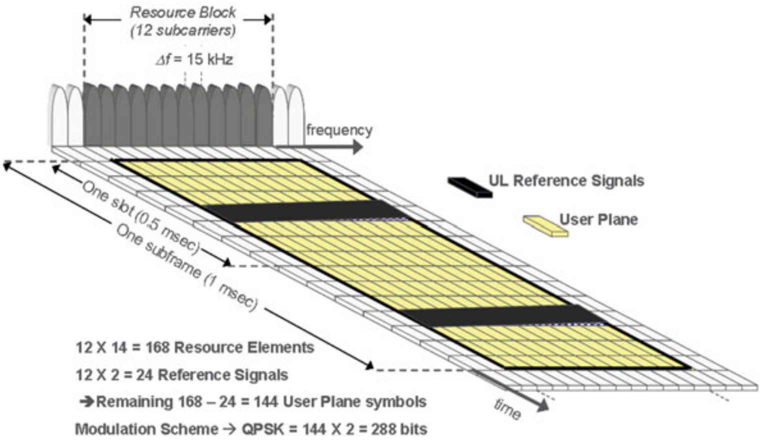

Fig. 3.16 Orthogonal frequency-division multiplexing (OFDM) user plane symbols for uplink transmission with quadrature phase shift keying (QPSK) action scheme

288 bits/ms. Regarding parameter n, using eNodeB-specific traffic counters from real traffic measurements, number of nonscheduled TTI intervals could be easily estimated. As an example, authors could propose Ericsson counters; following ratio *pmSessionTimeUe*/t_d, where *pmSessionTimeUe* is an Ericsson-based counter to measure the average total session service time of a UE considering transmissions and nonscheduled periods and t_d is the downloading measured time of a known size file from a server in a drive test. Average size of MAC packet M_{mac} has been already estimated in the previous steps. A good estimation of M_I is approximated using the ratio *PmPdcpVolUlDrb*/t_m, where *PmPdcpVolUlDrb* measures total uplink volume (PDCP SDUs) in an established Data Radio Bearer per measurement period in (kb) and t_m provides the measurement period. Considering both Eqs. (3.10) and (3.11), overall delay in the uplink transmission could be evaluated and compared with Fig. 3.10 3GPP QoS restrictions.

If during predefined step estimations, one step does not fulfill VoIP 3GPP standard restrictions or operator VoIP uplink determined requirements (throughput or BER), radio planner should reconsider (loose up some restrictions) cell size or path loss requirements and start recalculating cell size for smaller coverage conditions until all requirements and restrictions are fulfilled.

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